

**SONY®**

# ***Digital Dynamic Filter Plus***

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Operating Instructions  
Mode d'emploi

*DPS-F7*

# Warning

## FOR CUSTOMERS IN THE UNITED STATES

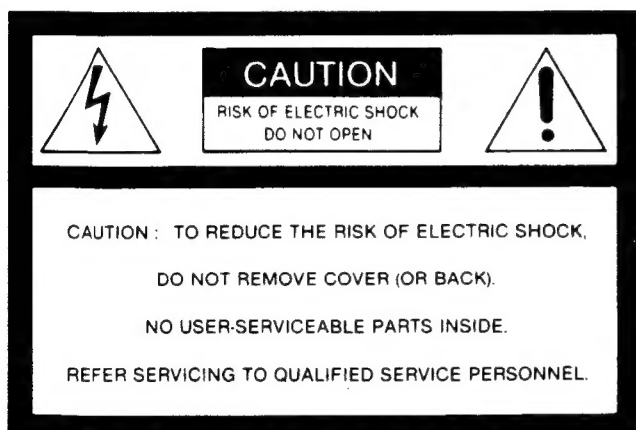
### Owner's Record

The model and serial numbers are located at the rear. Record these numbers in the space provided below. Refer to these numbers whenever you call upon your Sony dealer regarding this product.

Model No.    DPS-F7    Serial No. \_\_\_\_\_

## WARNING

**To prevent fire or shock hazard, do not expose the unit to rain or moisture.**



This symbol is intended to alert the user to the presence of uninsulated "dangerous voltage" within the product's enclosure that may be of sufficient magnitude to constitute a risk of electric shock to persons.



This symbol is intended to alert the user to the presence of important operating and maintenance (servicing) instructions in the literature accompanying the appliance.

For detailed safety precautions, see the leaflet "IMPORTANT SAFEGUARD".

## WARNING

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

## CAUTION

You are cautioned that any changes or modifications not expressly approved in this manual could void your authority to operate this equipment.

## FOR CUSTOMERS IN CANADA

### CAUTION

TO PREVENT ELECTRIC SHOCK, DO NOT USE THIS POLARIZED AC PLUG WITH AN EXTENSION CORD, RECEPTACLE OR OTHER OUTLET UNLESS THE BLADES CAN BE FULLY INSERTED TO PREVENT BLADE EXPOSURE.

This apparatus complies with the Class B limits for radio noise emission set out in Radio Interference Regulations.

# Precautions

FOR CUSTOMERS IN THE UNITED KINGDOM

## WARNING

THIS APPARATUS MUST BE EARTHED

## IMPORTANT

The wires in this mains lead are coloured in accordance with the following code.

Green-and-yellow:	Earth
Blue:	Neutral
Brown:	Live

As the colours of the wires in the mains lead of this apparatus may not correspond with the coloured markings identifying the terminals in your plug proceed as follows:  
The wire which is coloured green-and-yellow must be connected to the terminal in the plug which is marked by the letter E or by the safety earth symbol  $\perp$  or coloured green or green-and-yellow. The wire which is coloured blue must be connected to the terminal which is marked with the letter N or coloured black. The wire which is coloured brown must be connected to the terminal which is marked with the letter L or coloured red.

## On Safety

- To avoid electrical shock, do not open the cabinet. Refer servicing to qualified personnel only.
- Before connecting the unit to the power source, check to confirm that the operating voltage of your unit is the same as the local power line voltage. The operating voltage is indicated on the nameplate on the left side of the unit.
- Should anything fall into the cabinet, unplug the unit and have it checked by qualified personnel before operating it any further.
- The unit is not disconnected from the mains (AC power source) as long as it is connected to the mains outlet, even if the unit itself has been turned off.

## On Installation

- Allow adequate air circulation to prevent internal heat build-up; for example, when mounting this unit on the rack.
- Do not place the unit on a surface (rugs, blankets, etc.) or near materials (curtains, draperies, etc.) that may block the ventilation holes.
- Do not install the unit near heat sources such as radiators or air ducts or in a place subject to direct sunlight, excessive dust, mechanical vibration or shock.
- The unit is designed for operation in a horizontal position. Do not install it in an inclined position.
- Keep the unit away from equipment with strong magnets, such as microwave ovens or large loudspeakers.
- Do not place any heavy object on the unit.

## On Operation

- When the unit is not in use, turn the power off to conserve energy and to extend its life.

## On Cleaning

- Clean the cabinet, panel and controls with a dry soft cloth, or a soft cloth slightly moistened with a mild detergent solution.
- Do not use any type of solvents, such as alcohol or benzine, which might damage the finish.

## On Repacking

- Keep the carton and packing materials. They make an ideal container to transport the unit.

If you have any questions about the unit, contact your Sony service facility.

## CAUTION!

Danger of explosion if battery is incorrectly replaced. Replace only with the same or equivalent type recommended by the equipment manufacturer. Discard used batteries according to manufacturer's instructions.

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# Overview of the DPS-F7

*The DPS-F7 Digital Dynamic Filter Plus is a new signal effector using innovations in signal processing, based on digital filter technology. The ten types of algorithms create a superior sound environment considerably exceeding the possibilities available through conventional effectors.*

## Quality-conscious design with high-performance A/D and D/A converters

The DPS-F7 converts an incoming analog signal to a digital signal and outputs as an analog signal after passing it through various effect processes. The determining factor for the sound quality is the conversion mechanism which incorporates an 18-bit oversampling stereo A/D converter and a 49.152 MHz clock advanced pulse D/A converter, which together results in highly accurate effects with little deterioration of quality.

## User-friendly and comfortable operation

The large size backlit LCD of 40 characters by 2 lines enables smooth operation while viewing the operating condition in real time. The on-line manual (in English) can be displayed on the LCD so operation instructions are immediately available.

## Abundant preset memories

The DPS-F7, in its preset memory, has a hundred different effects created by musicians, sound mixers and acoustic engineers around the world. This will therefore enable you to select and replay the desired effects for a particular purpose immediately.

## Creation of any kind of sound

The EDIT function allows you to change the preset effects or create your original effects. Aside from the preset memory of 100 effects, the DPS-F7 also has a user memory in which up to 256 additional effects can be saved, giving quick access to an even greater variety of effects.

## Wide range of effects

The DPS-F7 is equipped with an effector which processes and output signals, a vocoder which modulates input signals to one channel according to those on the other channel, and a synthesizer which produces sounds by MIDI signals. In addition to these three fundamental functions, a newly developed algorithm, based on advanced digital filter technology, enables you easily to create a wide range of effects and synthesizer sounds.

It is also easy to modify the effects, since the effect block where the effects are modified is divided into sections, under which you can easily find the parameters you want to change.

## Remote control

A remote controller (RM-DPS7) is also available separately.

## Two I/O terminal systems

The DPS-F7 is equipped with XLR connectors (balanced type) and phone jacks through which you can connect musical instruments, recording equipment and PA (public address) equipment.

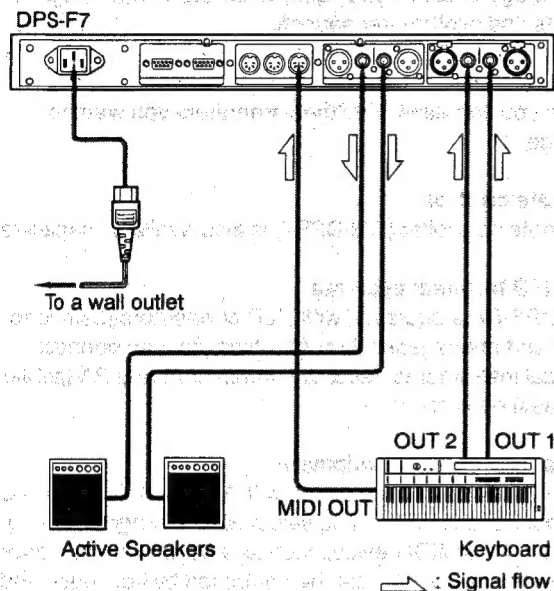
## Linkage with MIDI equipment

Since the DPS-F7 is equipped with MIDI functions, memory numbers of this unit can be selected with program change signals of the MIDI device, such as a keyboard. Moreover, since effect level, etc. can be controlled by key touch and control change signals, the unit is highly effective as an effector for digital musical instruments. Automatic performance is also possible by controlling with computers having the MIDI interface, or with a MIDI sequencer.

# Hooking Up a System

Turn all the power off before making connections, and connect the AC power cord last.

## Fundamental Connections



1. Connect a keyboard to the INPUT jacks, and the MIDI IN connector.
2. Connect active speakers to the OUTPUT jacks.
3. Insert the AC power cord firmly into the AC IN jacks.
4. Connect the AC power cord to a wall outlet.

### For the model equipped with a voltage selector

Check to confirm that the voltage selector is set to the local power line voltage. If not, set the selector to the correct position before connecting the AC power cord to a wall outlet.

### Notes:

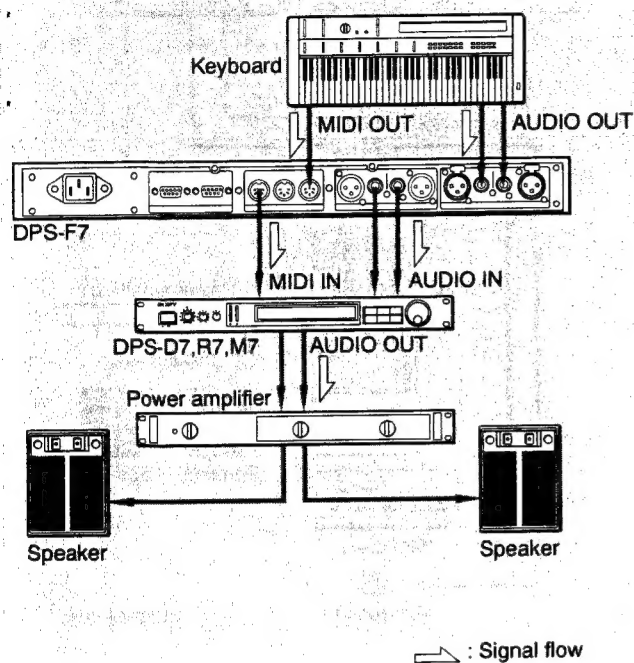
- Be sure to insert the plugs firmly into the jacks. Loose connection may cause hum and noise.
- Leave a little slack in the connecting cord to allow for inadvertent shock or vibration.
- Connections with some equipment of which the output capacity is very high may result in sound distortion. When this happens, turn the INPUT control to lower the input level of the equipment connected to the DPS-F7.

### Notes on input signals:

- The input mode of the system block is used to treat a variety of input signals. It can be set to either "stereo" or "mono." When it is set to "mono," input signals to channel 2 are ignored and those to channel 1 are sent to the effect block for both channels.
- Set the input mode to "stereo" for mono-in/mono-out type algorithm or stereo-in/stereo-out type algorithm. When using the mono-in/mono-out type algorithm, it is all right to use either channel as the input channel. However, some mono-in/mono-out algorithms are specially designed for channel 1. So, it is usually best to use channel 1 as the input channel.
- Set the input mode to "mono," for processing monaural signals separately for left and right channel, using different parameters for each channel.
- The reference level is fixed at  $-10$  dBs for the phone jacks and  $+4$  dBs for XLR type connector. The head margin is 20 dB. Therefore, the maximum input level for the phone jacks is  $+10$  dBs, and for XLR type connector  $+24$  dBs. To obtain the best possible S/N ratio, adjust the maximum input level either by adjusting the level of original signal before putting out to this unit or by adjusting the INPUT control on this unit.  
(0 dBs = 0.775 Vrms)

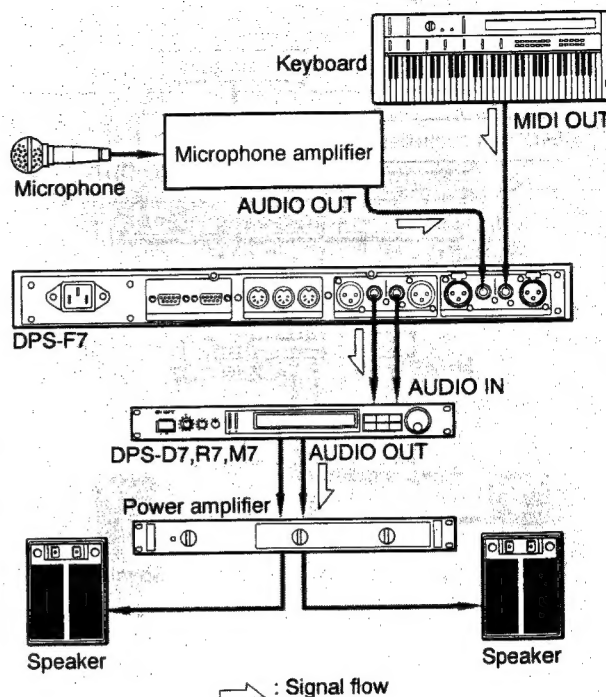
## Applications

### Hookup 1 – as an effector for the keyboard



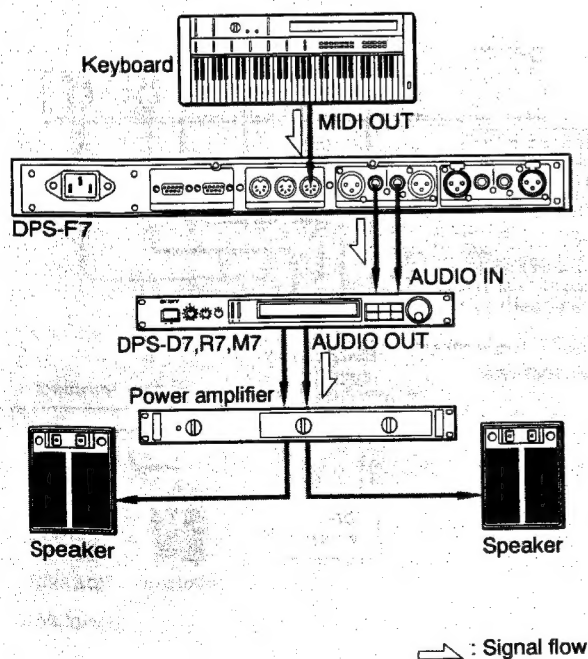
### Hookup 3 – as a vocoder

Audio output signals from the keyboard are modulated by signals input to the microphone.



### Hookup 2 – as a monophonic or percussion synthesizer

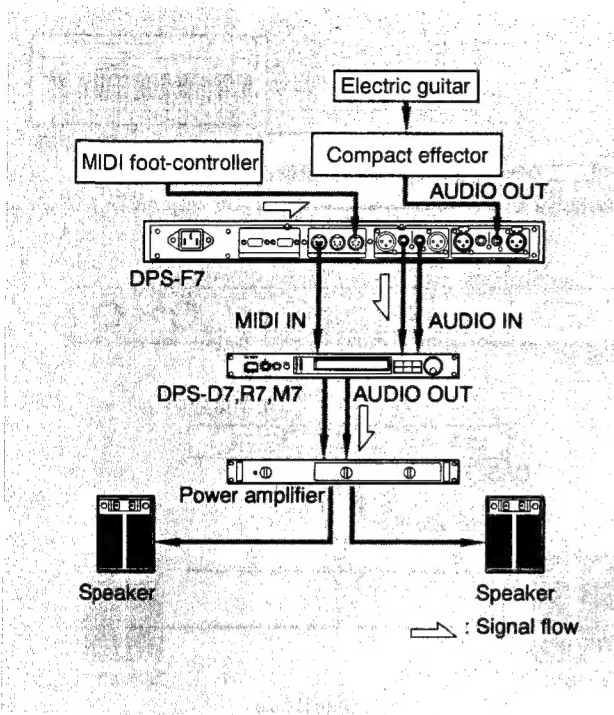
The DPS-F7 is used as a MIDI sound source module.



## Hooking Up a System

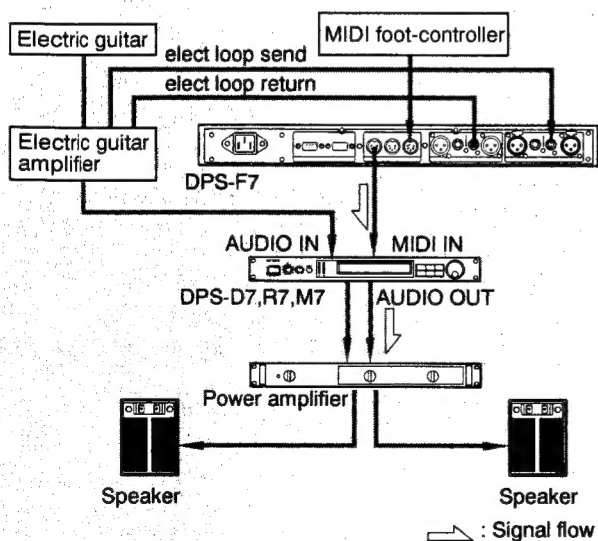
### Hookup 4.1 – as an effector for the electric guitar

A compact effector or direct box is connected in series.



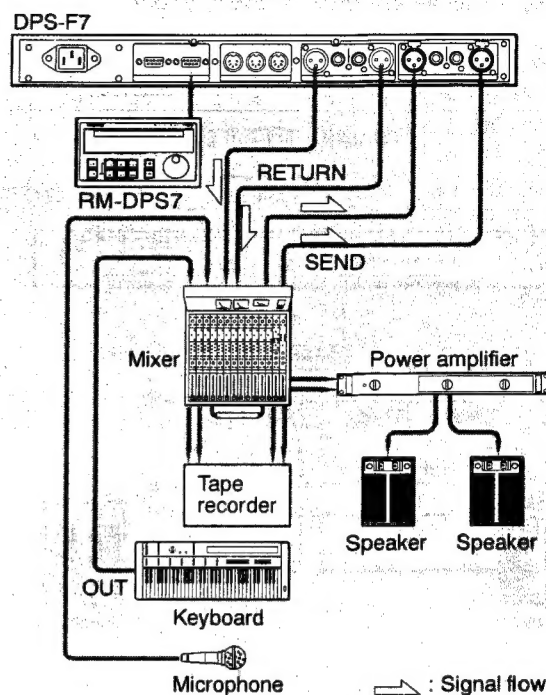
### Hookup 4.2 – as an effector for the electric guitar

The DPS-F7 is connected into the effect loop (accessory input/output) terminal of the guitar preamplifier or guitar amplifier.



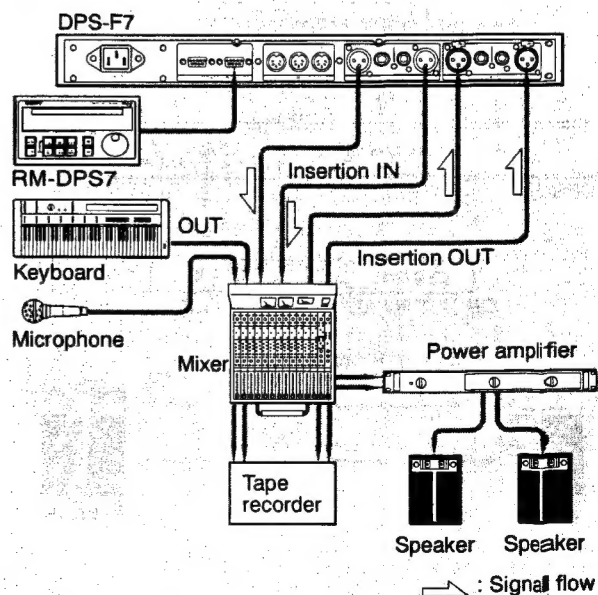
Set the input mode to "mono" when there are two effect loop return channels.

### Hookup 5 – for recording



### Hookup 6 – for recording

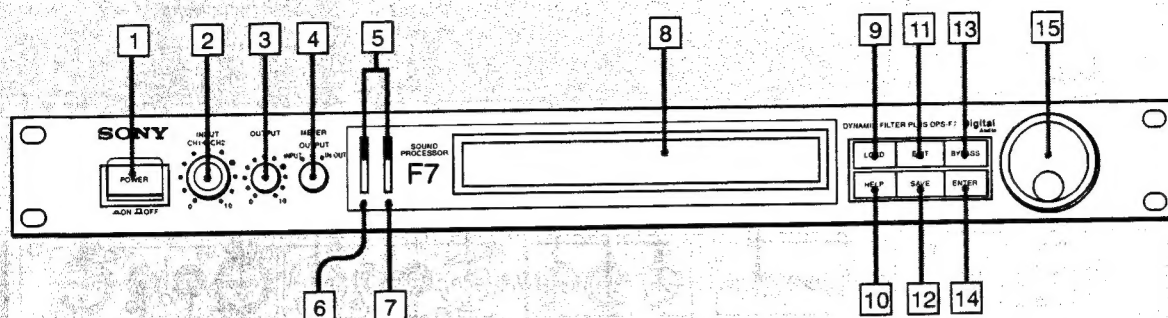
The insertion terminals for the channel you want to add effects to are used.



## Front Panel

English

Hooking up a System/Identifying the Parts



### 1 POWER switch

Turns the unit on and off. When the power is on, the display window lights up and whatever was last displayed appears again. For a few seconds after turning on the power, the sound being input will be output directly since the bypass function is on.

### 2 INPUT control

Adjusts the input level of individual channels. The outer control is for channel 1 and the inner control is for channel 2. Since the controls are linked, turn one control while holding the other still for adjustment of only one channel. Gain will become 0 dB if this control is turned up to the two o'clock position (largest point on the scale).

### 3 OUTPUT control

Adjusts the output level. Gain will become 0 dB if this control is turned up to the two o'clock position (largest point on the scale).

### 4 METER switch

Switches signals to be indicated on the level meter. If the switch is set to INPUT, the input signal level of each channel to the A/D converter will be indicated individually and, if set to OUTPUT, the output signal level of each channel from the D/A converter will be indicated individually. When this switch is set to IN/OUT, the channel 1 signal level being input to the A/D converter will show on the CH1 level meter and the level of the channel 2 signal output from the D/A converter will be shown on the CH2 level meter.

### 5 Level meter

Indicates the signal level. Adjust the INPUT control so 0 dB lights when a reference level signal is input. A 20 dB head room will be available when 0 dB lights. "OVER" will light if a signal exceeding the head room is input. The level meter remains inactive when the BYPASS button is pressed.

### 6 MIDI indicator

Lights when the MIDI program change signal or control change signal, etc. is received.

### 7 REMOTE indicator

Lights when a signal is received from an optionally available remote controller (RM-DPS7).

### 8 Display window

Memory names, parameter values and messages accessed are displayed on an LCD display of 40 characters by 2 lines. The backlighting lets you read the screen easily in dark halls and studios.

### 9 LOAD button

Press to access the memory.

### 10 HELP button

Press to display various information required for operation.

### 11 EDIT button

Press to change parameter values stored in memory.

### 12 SAVE button

Press when storing original effects created by changing parameter values in the user memory.

### 13 BYPASS button

Press when putting out input signals directly. Connects the input/output jacks with a relay. While the bypass function is active, signals do not flow through the A/D, D/A, and other internal circuits.

### 14 ENTER button

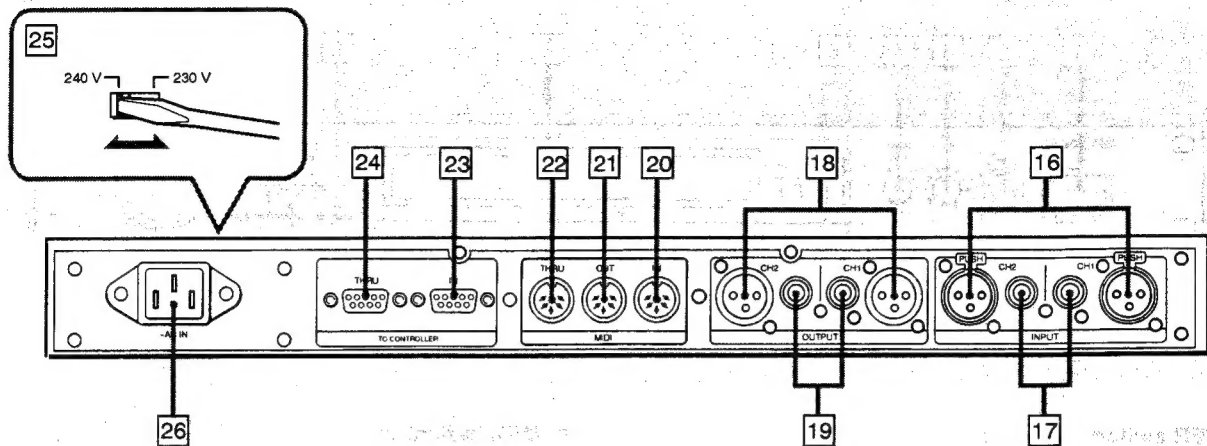
Press after selecting and setting parameters.

### 15 Operating dial

Selects preset numbers and/or sets parameters.

## Identifying the Parts

### Rear Panel



**16 INPUT CH1/CH2 terminal (XLR-3-31 connector)**  
Balanced-type terminals for input of channel 1 and channel 2.

**17 INPUT CH1/CH2 terminal (Phone jacks)**  
Phone jacks for input of channel 1 and channel 2.

**18 OUTPUT CH1/CH2 terminal (XLR-3-32 connector)**  
Balanced-type terminals for output of channel 1 and channel 2.

**19 OUTPUT CH1/CH2 terminal (Phone jack)**  
Phone jacks for output of channel 1 and channel 2.

\* When devices are connected to both XLR input connectors and phone input jacks, the device connected to the phone jacks will have priority.

**20 MIDI IN terminal (DIN 5-pin)**  
Input terminal for the MIDI signal. For connection to the MIDI OUT (or THRU) terminal of another MIDI device by means of a commercially available MIDI cable.

**21 MIDI OUT terminal (DIN 5-pin)**  
Outputs the MIDI signal generated by this unit.

**22 MIDI THRU terminal (DIN 5-pin)**  
Outputs MIDI signals input from the MIDI IN terminal as is, and can be connected to the MIDI IN terminal of a MIDI device with a commercially available MIDI cable.

**23 TO CONTROLLER IN terminal (D-Sub 9-pin)**  
Terminal to which the remote controller RM-DPS7 (not supplied) is connected to permit remote control of panel operation of the DPS-F7.

**24 TO CONTROLLER THRU terminal (D-Sub 9-pin)**  
Outputs directly the remote controller signals input from the TO CONTROLLER IN terminal. Connect to the TO CONTROLLER IN terminal of another effector in the DPS series.

**25 VOLTAGE SELECTOR**  
(only for UK and European model)  
Set the voltage selector to the correct position before connecting the AC power cord to a power outlet.

**26 AC IN terminal**  
Use the supplied AC power cord and connect it to an AC power outlet.

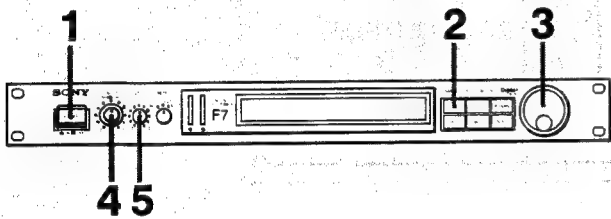
# Chapter 2 Operation at a Glance

## Let's Operate Your DPS-F7

The DPS-F7 has a hundred effects stored in its preset memory. Let's listen to these effects one by one, referring to "Hooking Up a System" (page 6) and "Preset Memory List" (separate volume).

Note that memory names used in this manual may be different from those actually displayed on your DPS-F7, and that these differences result from specification changes to give you the best possible effects.

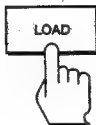
### Selecting a Preset Memory



1. Turn on the power.



2. Press the LOAD button.



LOAD mode indication  
(P=Preset memory, U=User memory)

Algorithm name	Memory name
11:PEM	Preset equalizer >A.guitar line REC 1

3. Turn the operating dial and select the desired preset number (P1 – P100, U1 – U256).



- For the contents of the preset memories (from P1 to P100), refer to the "Preset Memory List" (separate volume).
- Only user memories (from U1 to U256) which have been stored can be selected. (See page 58)

4. Turn the INPUT control to adjust the input level.



5. Turn the OUTPUT control to adjust the output level.



### Before turning on the connected devices

Lower the volume of each device to prevent unexpected loud sounds.

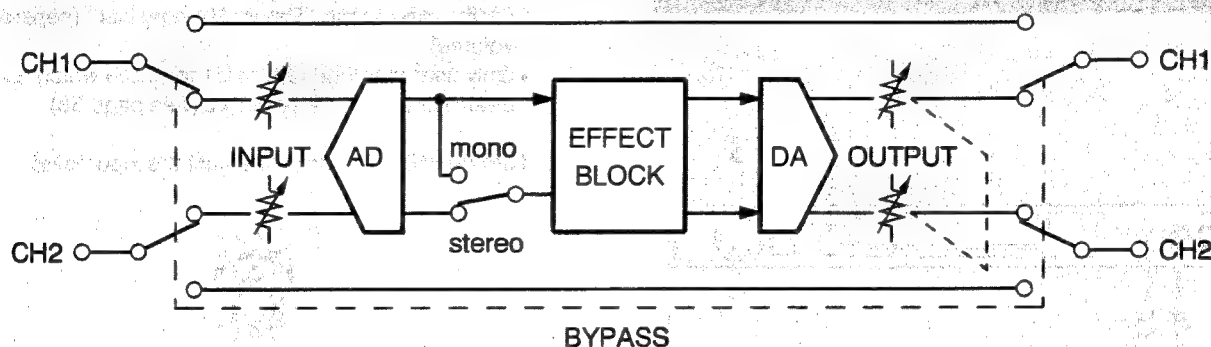
### To output the input signals directly

Press the BYPASS button to send the input signal directly to the output without any changes created by the effector. This function allows you compare the sound before and after you apply an effect. To release the bypass function, press the button once again.

# Overview of the Signal Processing Blocks

The signal digitalized by the A/D converter is processed sequentially in the effect block and the results are sent to the D/A converter.

## General Block Diagram



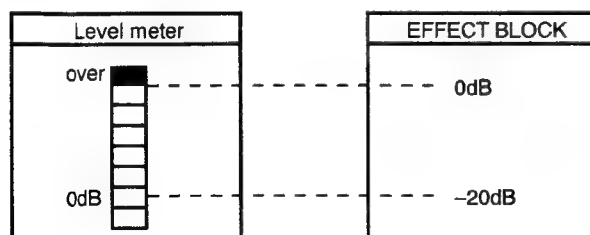
### Notes

- When using the bypass function, signals input to channel 1 and channel 2 bypass the internal circuits and are output directly to the output terminal. When switching off the unit, the bypass function is automatically on.
- The analog signal processing section has a gain of 10 dB for each input and output. Adjust the input and output levels with the INPUT and OUTPUT controls to make the I/O level suitable for the equipment connected to the DPS-F7. Setting the controls to the two o'clock position (largest point on the scale) produces a gain of approximately 0 dB.

### Relation between the level meter and internal signal level of the effect block

You can monitor, by using the level meter on the panel, the reference level (0 dB) and the maximum level (+20 dB) of analog signals which go in or come out from the digital processing block.

The maximum level is set at +20 dB for the level meter. On the other hand, it is set at 0 dB in the effect block. The diagram below explains the relation.



## Functions of Each Block

Only the effect block can influence signal processing. The other blocks are used to set the operating environments of the DPS-F7.

### Effect Block

Is used to obtain actual effects by setting the parameter values using the LOAD, EDIT, and SAVE buttons.

### Memory Block

Is used in editing the user memory.

### System Block

Is used to specify the operating environment of the DPS-F7.

### SYS. SG Block

Is used to specify the special operating environment related to algorithms "Percussion Synthesizer (PRC)" and "Monophonic Synthesizer (SYN)."

### SYS. MIDI Block

Is used to specify the MIDI operating environment of the DPS-F7.



# MIDI Controls

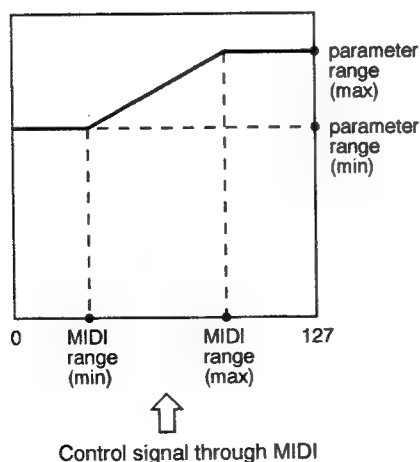
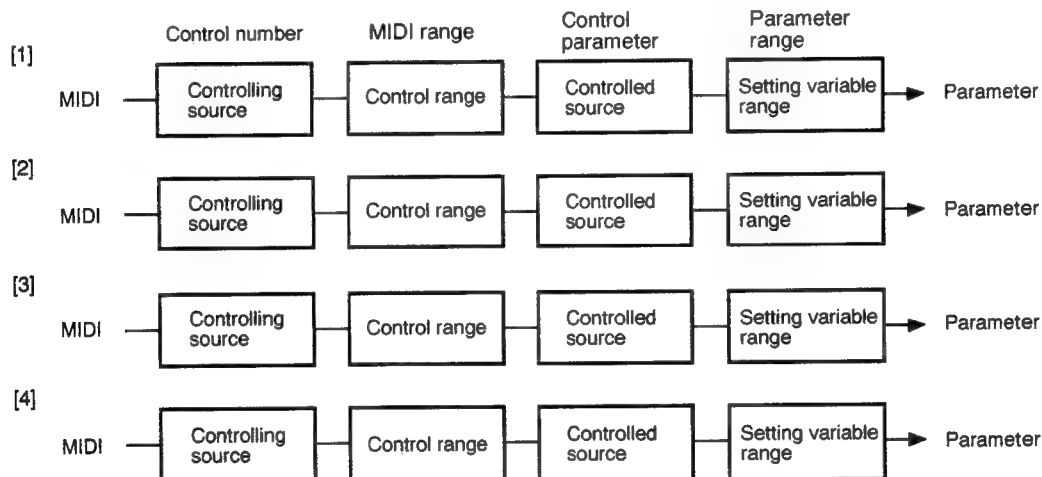
You can control the DPS-F7 through the MIDI interface. The basic settings for MIDI (MIDI channel, etc.) are set in the SYS. MIDI block. (See page 51.)

Internal memory numbers (or BYPASS) of this unit, each corresponding to the MIDI program change number (1 to 128), are to be assigned for the MIDI program change since the DPS-F7 has a large storage capacity for preset and user memory. This assignment is performed in the SYS. MIDI block.

The effect depth can be controlled in real time according to the MIDI control change data such as the damper and modulation wheel. These values are set in each memory because the MIDI control environment varies according to the algorithm used in the memory. The values are set in the L.MIDI section in the effect block. The DPS-F7 can accept four systematic controls through the MIDI interface, and for each of the four the following is to be determined.

- Controlling source (what MIDI data controls?)
- Control ranges (for both the controlling and controlled sources)
- Controlled source (what parameter is controlled?)

Algorithms, except for "Dynamic Filter 1 (DF1)" and "Dynamic Filter 2 (DF2)" are basically structured as shown in the following figure. See the corresponding explanation pages for algorithms which have other structures.



## Notes

- When the minimum and maximum values are identical
  - Increasing the minimum values also increases the maximum value.
  - Decreasing the maximum values also decreases the minimum value.
- Noise may be generated, depending on parameter settings by MIDI control.

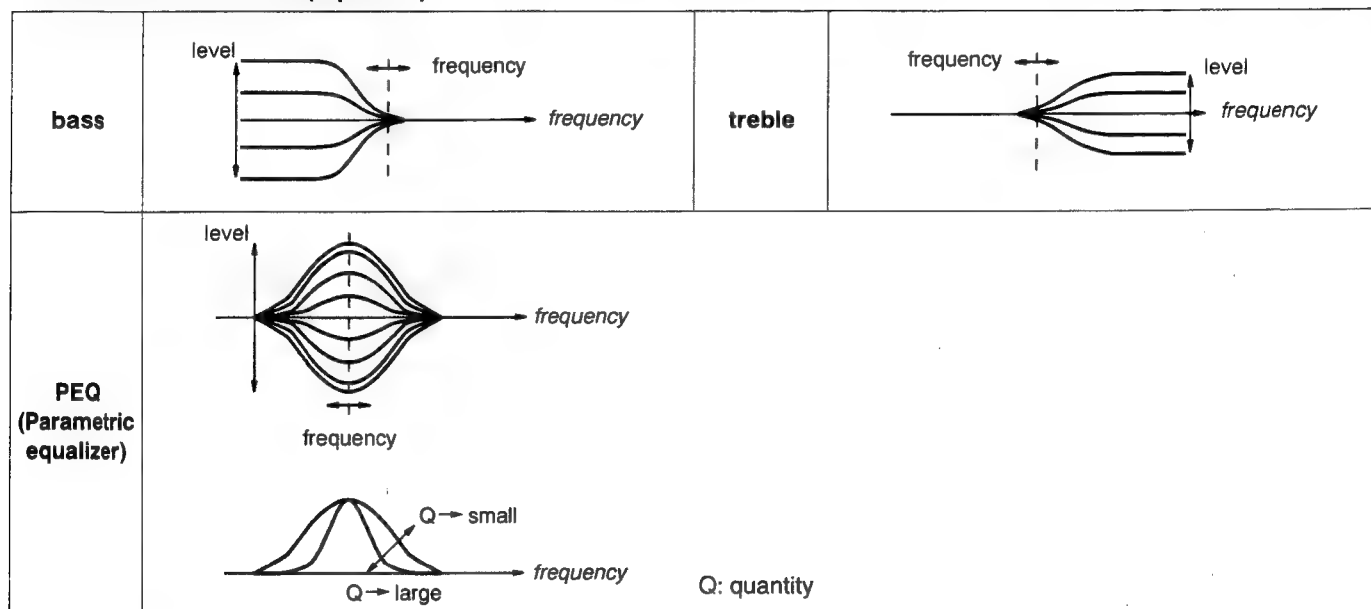
# Effect Block

You can adjust the sound only in the effect block. There are ten types of algorithms in which you can edit a variety of parameters.

## Common Parameters

This section describes and illustrates parameters commonly used in most of the algorithms.

### Parameters related to EQ (equalizer)



### soft clipper

The distortion which occurs when the signal is over the maximum level is reduced by rounding the amplitude response near the maximum level. (Figure I)

You can obtain the maximum output level by setting the "gain recover" parameter to "on" even when soft clip processing is performed. (Figure II)

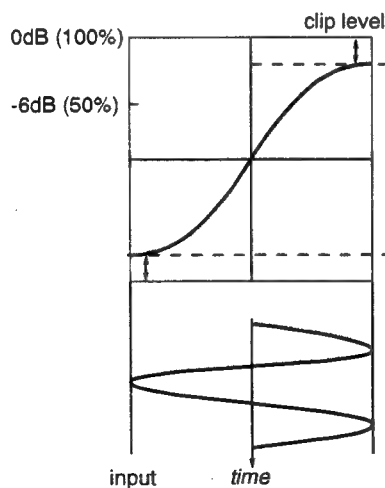


Figure I

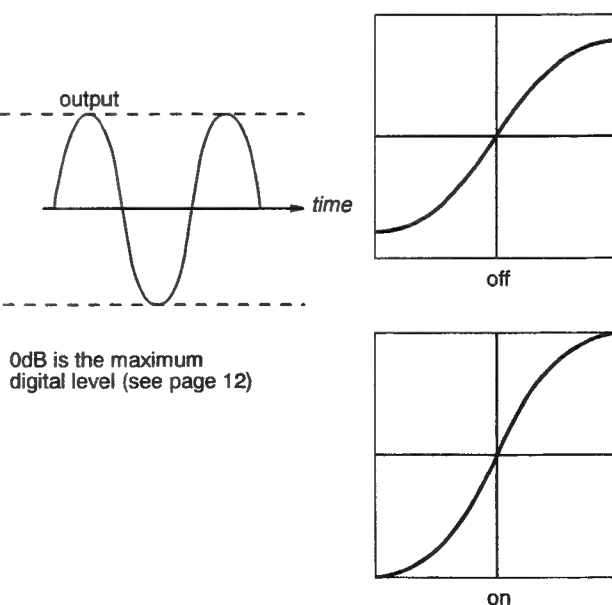


Figure II – gain recover explanation

**envelope follower (EF)/gate**

An *envelope* is a curve drawn by connecting the signal peak points (sound strength), and an *envelope follower* is a circuit that extracts an envelope from the signal. (Figure I)

The form of an envelope can be changed according to the parameter settings. (Figures IV, V, VI)

Gates used in an algorithm are, then, controlled with the envelopes. So, refer to the following figures when setting the parameters for the gate section.

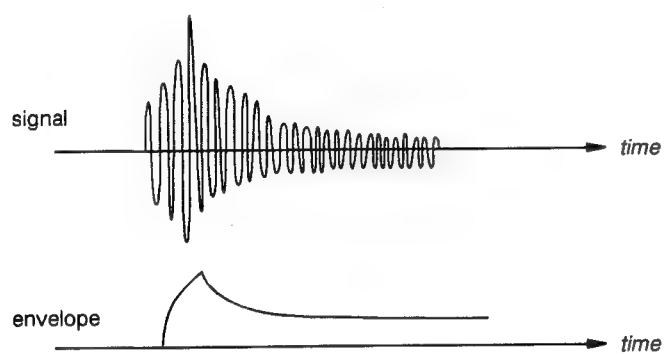


Figure III – envelope follower

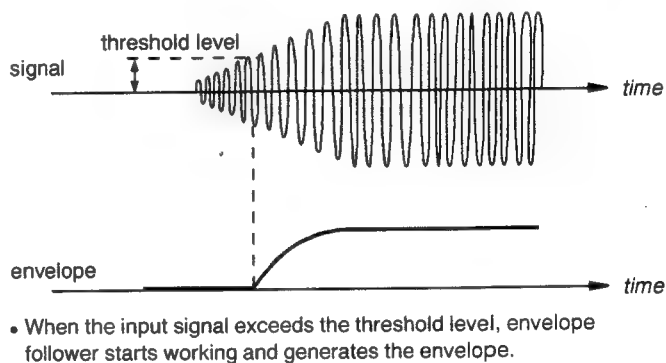


Figure V – threshold level explanation

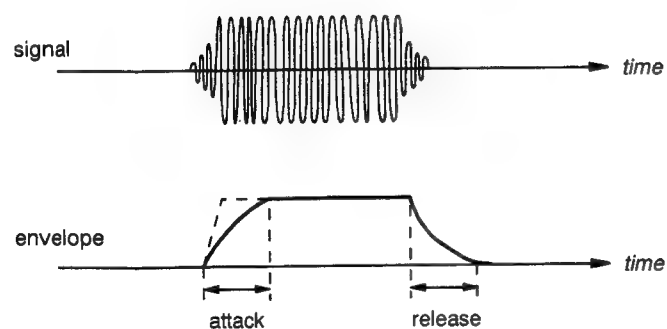


Figure IV – attack/release explanation

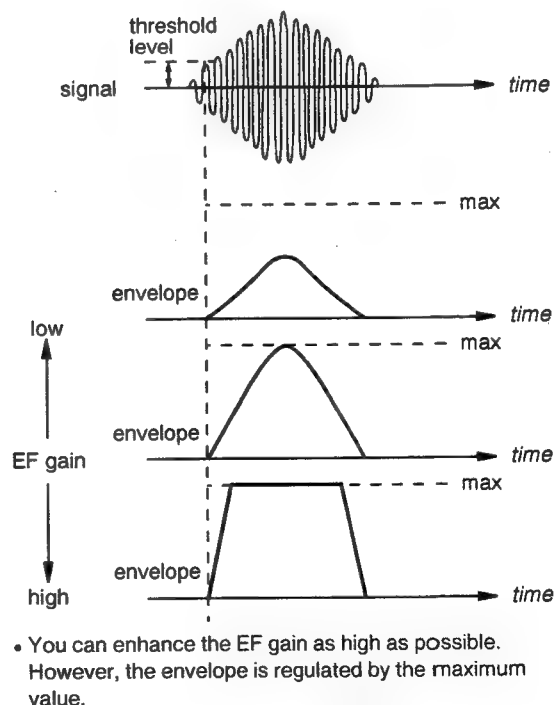


Figure VI – gain explanation

## Effect Block

### Utility Section

This section has two special functions: "section copy" and "section initialize." Unlike other sections, this section has no values to be stored.

The "section copy" parameter allows you to copy all the parameters within a specified section from the same algorithm to the temporary buffer. On the other hand, the "section initialize" parameter allow you to initialize a specified section.

#### Examples of "section copy"

##### Cans:

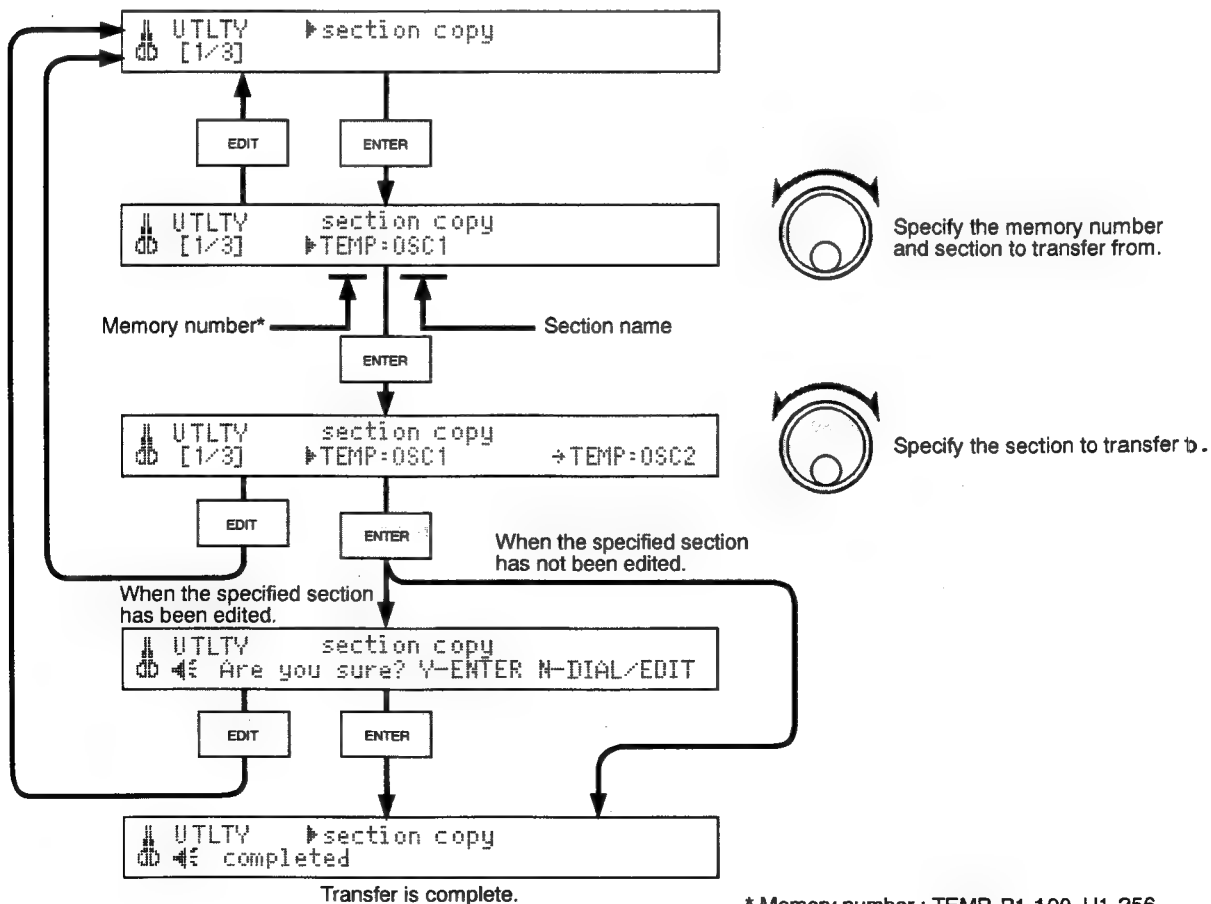
From			To		
Memory	Algorithm	Section	Memory	Algorithm	Section
Temporary buffer	[PEQ]	(PEQ1)	Temporary buffer	[PEQ]	(PEQ2)
Uxxx	[PEQ]	(PEQ1)	Temporary buffer	[PEQ]	(PEQ2)
Pxxx	[ECL]	(PEQ)	Temporary buffer	[ECL]	(PEQ)

##### Cannots:

From			To		
Memory	Algorithm	Section	Memory	Algorithm	Section
Pxxx	[PEQ]	(PEQ1)	Temporary buffer	[ECL]	(PEQ)
Pxxx	[PEQ]	(BASS)	Temporary buffer	[PEQ]	(TRBL)
				Attribution of section is different.	

#### "section copy" flow chart

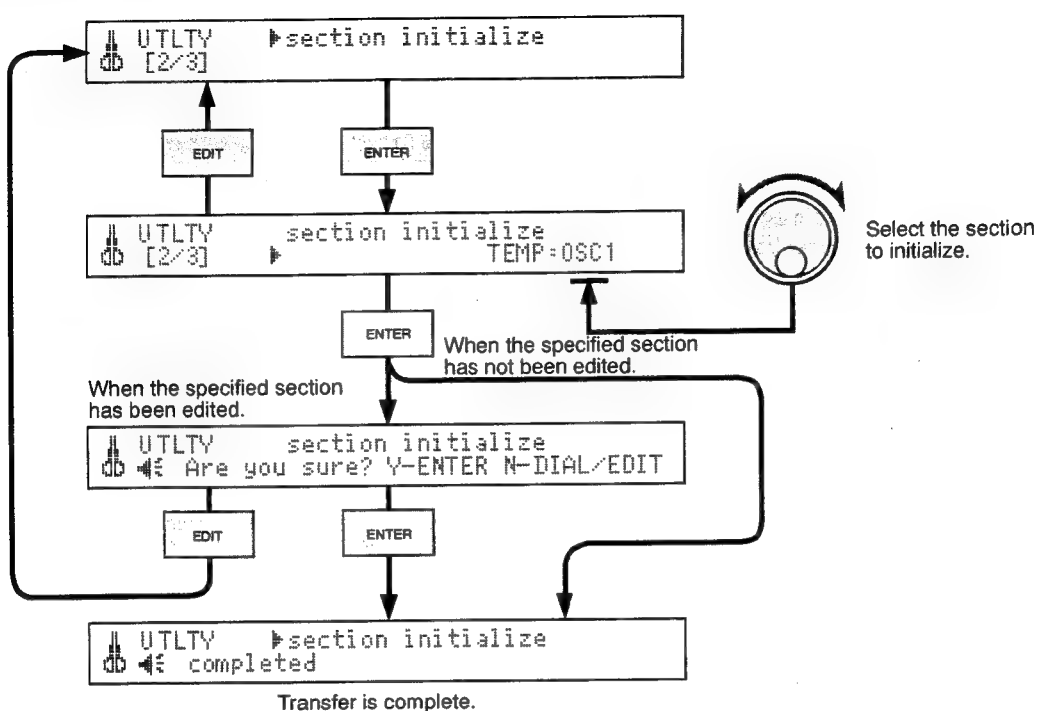
[section copy]



\* Memory number : TEMP, P1-100, U1-256

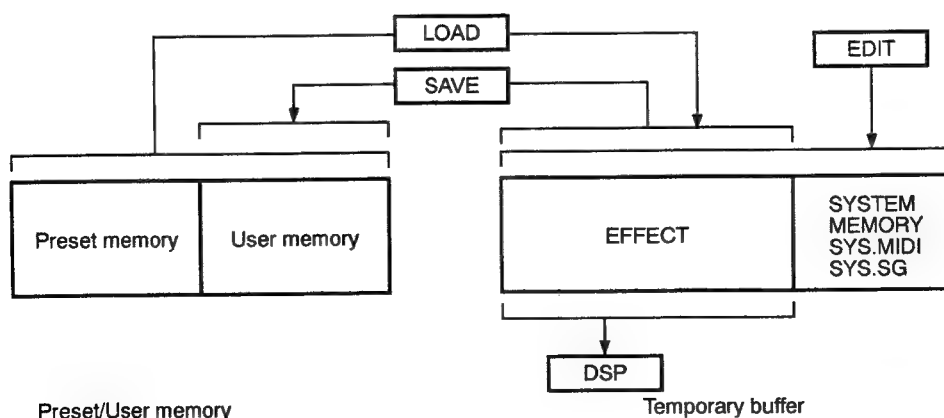
## "section initialize" flow chart

[section init]



## Relation between memory and temporary buffer

To copy a preset memory and a user memory data in the temporary buffer is called "to load." To edit the data in the temporary buffer and to write the results in the user memory is called "to save." Data in the temporary buffer is transferred to the DSP (Digital Signal Processor) and is reflected in the sound.



## Algorithm 0

Effect Off

OFF

The effect block is bypassed so that no effect can be obtained.

## Algorithm 1

Fading Parametric Equalizer

PEQ

This algorithm consists of a shelving type equalizer (bass, treble), a high-frequency adjuster that controls the ultra-high band frequency response for the sound quality fine adjustment, and four PEQs (parametric equalizers). This can be used as a MIDI automation equalizer because all parameters, except for SCLP section, change gradually for the program change between memories in which this algorithm is used.

### Section/Parameter Description

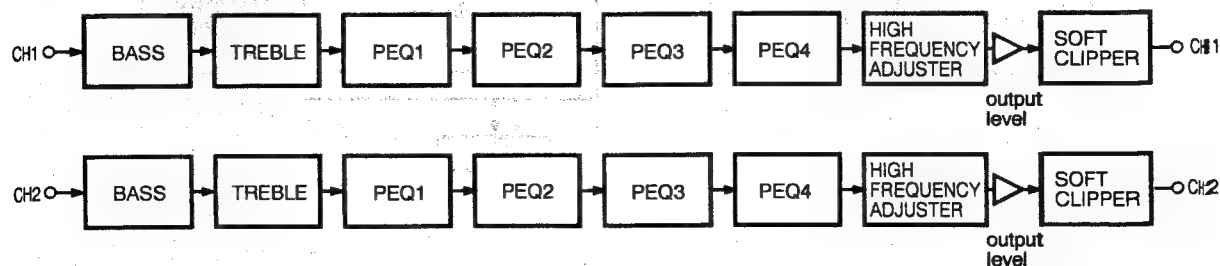
#### HFA (high frequency adjuster)

This is the exclusive equalizer for ultra-high band response compensation above 20 kHz.

#### SCLP (soft clipper)

Refer to "Common Parameters" on page 14.

SECTION	PARAMETER	MIN and MAX
BASS	bass on/off bass frequency (ch1, ch2) bass level (ch1, ch2)	on/off 16Hz – 6.3kHz – 12 – + 12 dB
PEQ1 – 4	PEQ1 – 4 on/off PEQ1 – 4 frequency (ch1, ch2) PEQ1 – 4 level (ch1, ch2) PEQ1 – 4 q (ch1, ch2)	on/off 63Hz – 18.0kHz – 12 – + 12dB 0.267/0.667/ 1.414/2.145/ 4.319/8.651/ 17.31
TRBL	treble on/off treble frequency (ch1, ch2) treble level (ch1, ch2)	on/off 400Hz – 20.0kHz – 12 – + 12dB
HFA (high frequency adjuster)	high freq adjuster on/off high freq adjuster level (ch1, ch2)	on/off – 12 – + 12dB
SCLP (soft clipper)	soft clipper on/off soft clipper level (ch1, ch2) gain recover (ch1, ch2)	on/off 0/ – 2/ – 4/ – 6dB on/off
OUTPUT	output level (ch1, ch2)	– ∞ – 0dB
L.MIDI	[1] – [4] control number  [1] – [4] MIDI range (min) [1] – [4] MIDI range (max) [1] – [4] control parameter  [1] – [4] parameter range (min) [1] – [4] parameter range (max)	0 – 31, 64 – 120: control change #0 – #31, #64 – #120/ note on velocity/ channel pressure/ note number (note on)/off  0 – 127 0 – 127 all (except on/off parameter) depending on parameter depending on parameter



Block diagram

This algorithm lowers the signal input level at the input stage to  $-18$  dB to avoid clipping at the equalizing stages. The level setting for the entire equalization is performed by the "output level" parameter. When the "output level" parameter is set to  $0$  dB, a gain of  $+18$  dB is added at the output stage and the total gain comes to  $0$  dB when the equalizers are all "off." Adjust the "output level" parameter when clipping occurs at the boost setting. (Figure 1-1)

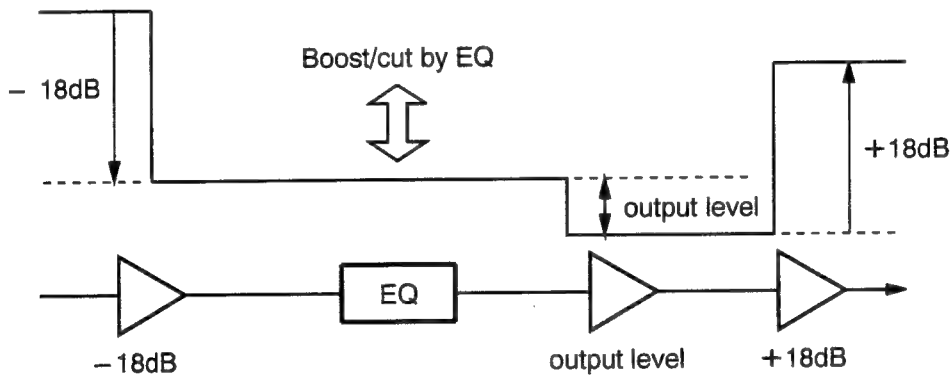


Figure 1-1 – EQ level diagram

## Algorithm 2 Dynamic Filter 1

DF1

The levels of two PEQs and of the HFB (high frequency booster), LPF (Low-pass filter) cutoff frequency and LPF maximum attenuation level are controlled by the independent envelope followers or MIDI signals. You can create such effects as enhancement or noise reduction according to your own needs. It is usually difficult to create the best possible effects, related to the envelope, and so requires a sort of trial-and-error approach. However, the introduction of "initial" and "target" parameters allows you somewhat appropriate editing.

SECTION	PARAMETER	MIN and MAX
INPUT	input level (ch1, ch2)	-18 - 0dB
OUTPUT	output level (ch1, ch2)	-12 - +12dB
PEQ1, 2	PEQ on/off mode frequency q  level modulation source initial level  target level	on/off boost/cut 63Hz - 20kHz 0.267/0.667/1.414/ 2.145/4.319/8.651/ 17.31 off/[1]~[4] MIDI/EF (boost) 0 - +18dB (cut) -18 - 0dB (boost) 0 - +18dB (cut) -18 - 0dB
P1, 2-EF (PEQ1, 2 envelope follower)	EF source EF sidechain filter side filter frequency side filter q  side PEQ level threshold level attack time release time EF gain	ch1/ch2/mix/panel key off/PEQ/BPF 63Hz - 20kHz 0.267/0.667/1.414/ 2.145/4.319/8.651/ 17.31 -12 - +12dB -72 - 0dB 0 - 489.6msec 0 - 7.83sec 0 - +60dB
LPF	LPF on/off freq modulation source initial frequency target frequency level modulation source initial stopband level target stopband level	on/off off/[1]~[4] MIDI/EF 16Hz - 20kHz 16Hz - 20kHz off/[1]~[4] MIDI/EF -∞ - 0dB -∞ - 0dB
LPF-EF (LPF envelope follower)	EF source EF sidechain filter side filter frequency side filter q  side PEQ level threshold level attack time release time EF gain	ch1/ch2/mix/panel key off/PEQ/BPF 63Hz - 20kHz 0.267/0.667/1.414/ 2.145/4.319/8.651/ 17.31 -12 - +12dB -72 - 0dB 0 - 489.6msec 0 - 7.83sec 0 - 60dB
HFB (high frequency booster)	high freq booster on/off filter slope level modulation source initial level target level	on/off 1/2/3 off/[1] - [4] MIDI/EF 0 - +18dB 0 - +18dB

SECTION	PARAMETER	MIN and MAX
HFB-EF (HFB envelope follower)	EF source  EF sidechain filter side filter frequency side filter q  side PEQ level threshold level attack time release time EF gain	ch1/ch2/mix/ panel key off/PEQ/BPF 63Hz - 20kHz 0.267/0.667/ 1.414/2.145/ 4.319/8.651/ 17.31 -12 - +12dB -72 - 0dB 0 - 489.6msec 0 - 7.83sec 0 - 60dB
SCLP (soft clipper)	soft clipper on/off soft clipper level gain recover	on/off 0/-2/-4/-6dB on/off
L.MIDI	[1] - [4] control number    [1] - [4] MIDI range (min) [1] - [4] MIDI range (max)	0 - 31, 64 - 120: control change #0 - #31, #64 - #120/ note on velocity/ channel pressure/ note number (note on)/off 0 - 127 0 - 127

### Note

Control range by MIDI signals is also subject to the "initial" and "target" parameter settings.

### Section/Parameter Description

#### EF source

This parameter obtains the maximum value (that is, the target value is reached) when you press the ENTER button with "panel key" selected.

#### HFB (high frequency booster)

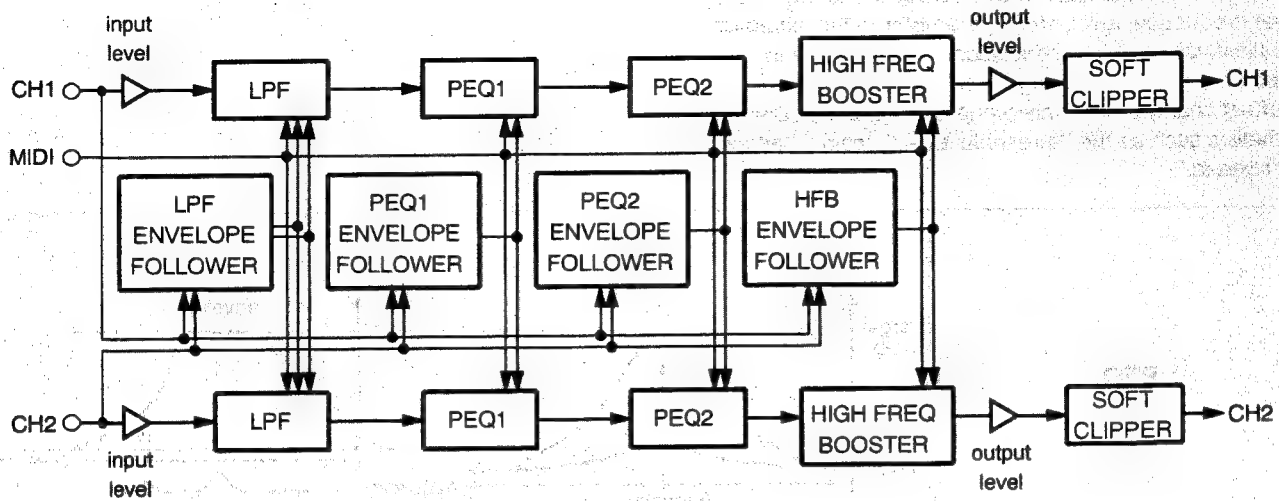
This section is a linear phase equalizer only for high frequency band.

#### EF sidechain filter

This is an equalization filter only for envelope follower input.

#### side PEQ level

This parameter is valid only when "PEQ" is selected in the "EF sidechain filter" parameter.



Block diagram

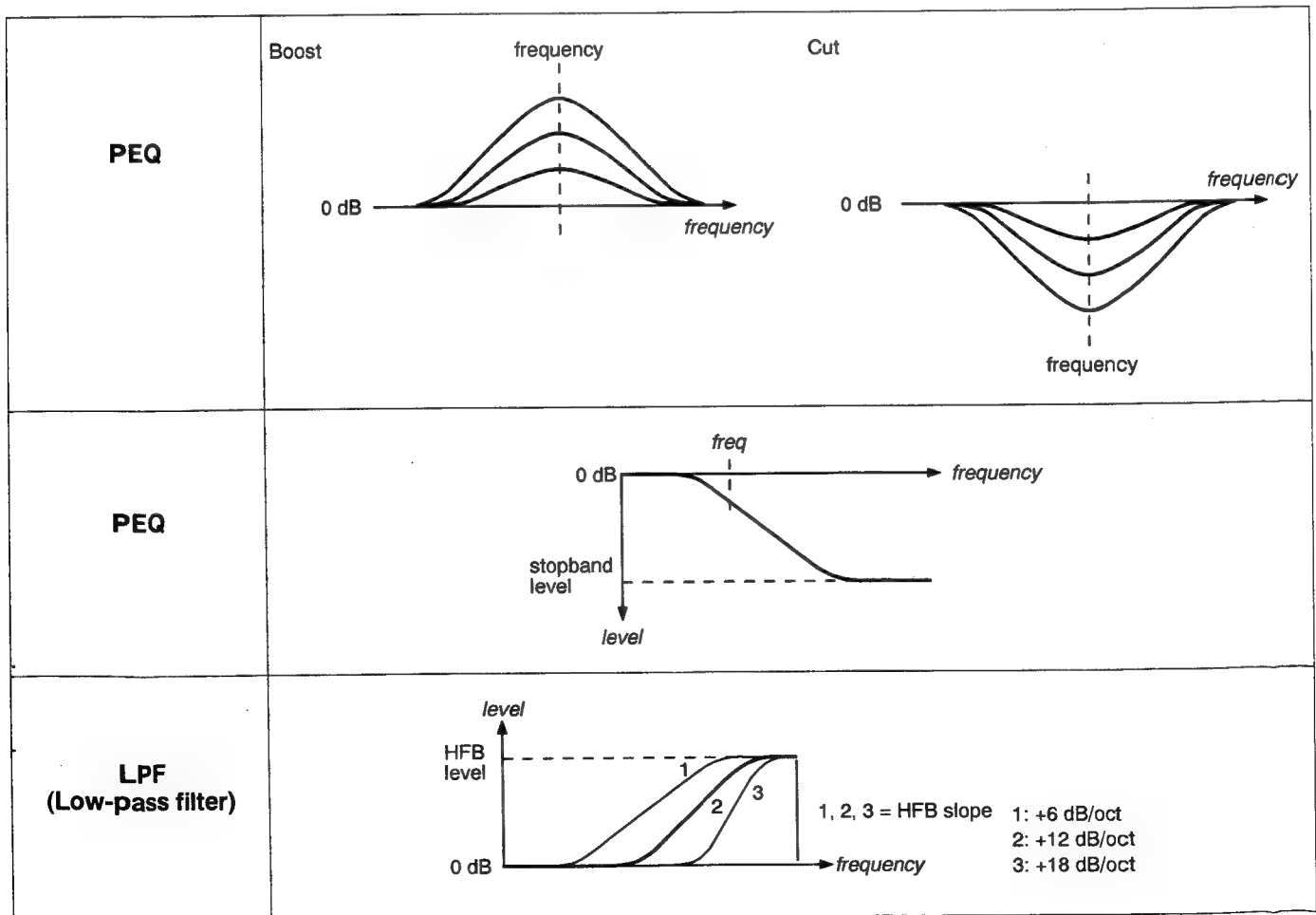


Figure 2-1 – Filter characteristics

## Effect Block

### Initial and Target Settings

Set the "initial (frequency, level, etc.)" parameter to a given value, which no modulation would be applied to. Set the "target (frequency, level, etc.)" parameter to the expected value (limit value) which modulation would have been applied to. The variation range is specified by these settings. The effect change within this range is set then by other parameters such as the "threshold level," "gain," "attack," and "release."

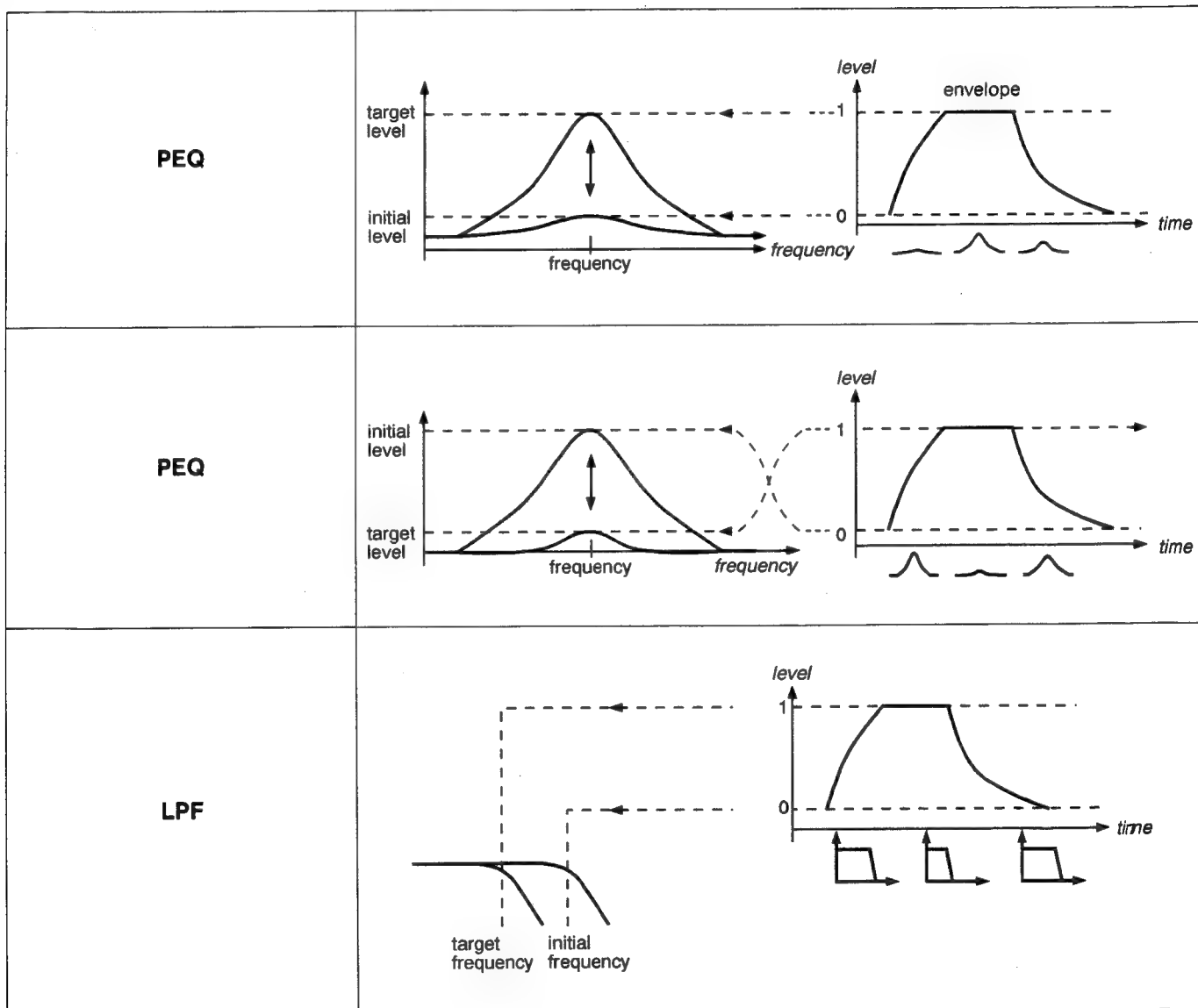


Figure 2-2 – Initial and target control explanation

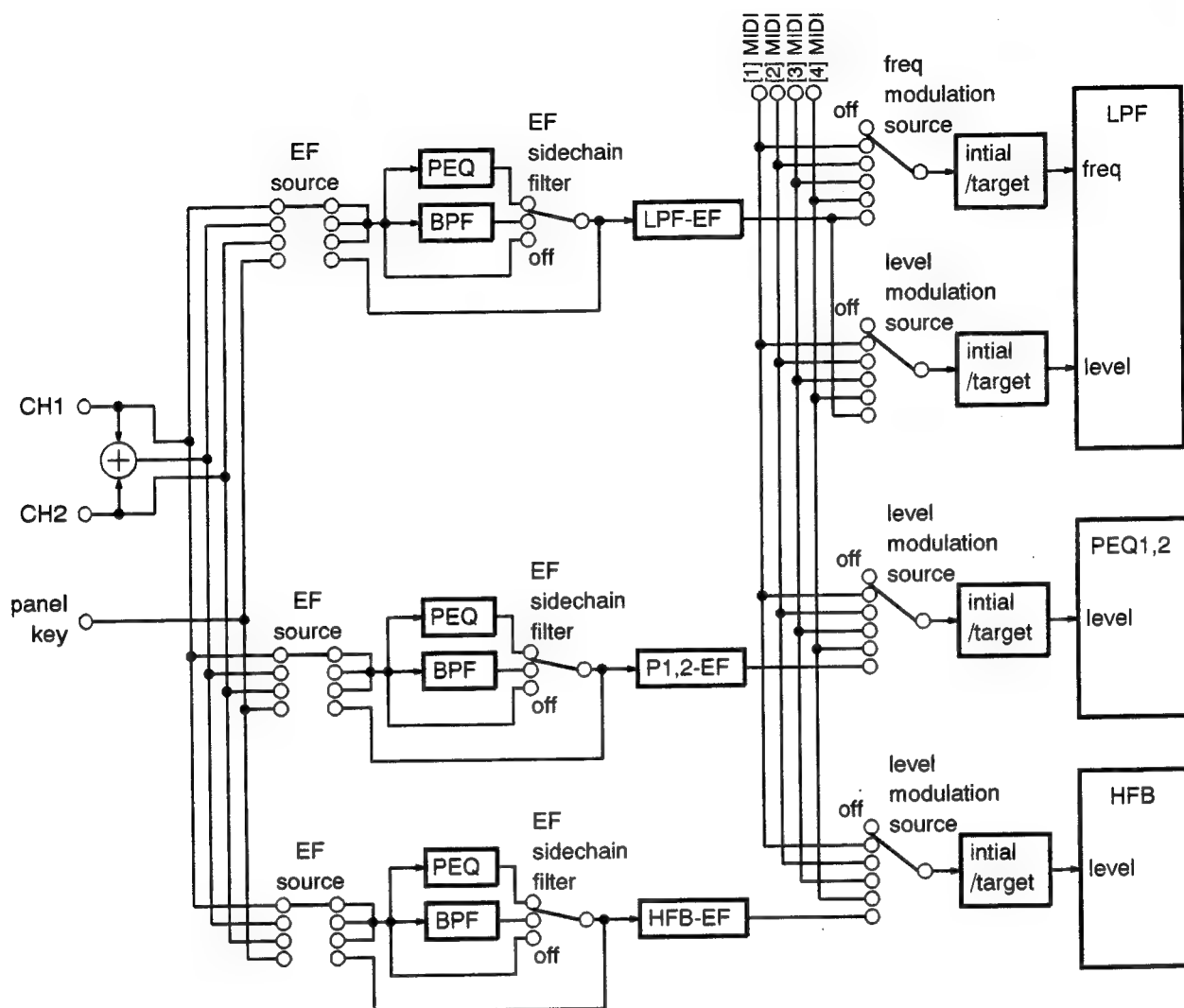


Figure 2-3 – Control system explanation

## Algorithm 3 Dynamic Filter 2 DF2

This is a variable dynamic filter which can be controlled by the outputs of the envelope follower and of the envelope generator. This dynamic filter consists of MMF (Multi-mode filter) and BPF (Bandpass filter) units in which all elements are variable, a fixed HPF (High-pass filter) and a DCF (Dynamic cross fader). The dynamic cross fader enables cross fade between two systematic input signals (see Figure 3-1). So, you can smoothly switch signals of different frequency response. Real time control of this algorithm allows you to change a plural number of elements simultaneously from only one control source.

SECTION	PARAMETER	MIN and MAX
OUTPUT	output level (ch1, ch2)	0 – 100%
BPF	BPF on/off	on/off
	BPF frequency	0 – 100
	BPF freq modulation source	off/[1] – [4] MIDI/EF1/LFO/EG
	BPF freq mod intensity	0 – 100
	BPF freq modulation phase	normal/inverse
	BPF level	0 – 400%
	BPF level modulation source	off/[1] – [4] MIDI/EF1/LFO/EG
	BPF level mod intensity	0 – 100
	BPF level modulation phase	normal/inverse
	BPF q	0 – 100
	BPF q modulation source	off/[1] – [4] MIDI/EF1/LFO/EG
	BPF q modulation intensity	0 – 100
	BPF q modulation phase	normal/inverse
MMF (multi mode) filter	MMF on/off	on/off
	MMF filter mode	BPF/LPF/HPF
	MMF frequency	0 – 100
	MMF freq modulation source	off/[1] – [4] MIDI/EF1/LFO/EG
	MMF freq mod intensity	0 – 100
	MMF freq modulation phase	normal/inverse
	MMF level	0 – 400%
	MMF level modulation source	off/[1] – [4] MIDI/EF1/LFO/EG
	MMF level mod intensity	0 – 100
	MMF level modulation phase	normal/inverse
	MMF q	0 – 100
	MMF q modulation source	off/[1] – [4] MIDI/EF1/LFO/EG
	MMF q modulation intensity	0 – 100
	MMF q modulation phase	normal/inverse
HPF	HPF on/off	on/off
	HPF frequency	16Hz – 20kHz
	HPF phase	normal/inverse
DCF (dynamic cross fader)	mix level (HPF: BPF + MMF)	100%:0% – 0%:100%
	modulation source	off/[1] – [4] MIDI/EF2/LFO
	modulation intensity	0 – 100%
	modulation phase	normal/inverse
EF1 (envelope follower 1 for filter)	EF source	ch1/ch2/mix/panel key
	EF sidechain LPF frequency	16Hz – 20kHz
	EF sidechain HPF frequency	16Hz – 20kHz
	threshold level	– 72 – 0dB
	attack time	0 – 489.6msec
	release time	0msec – 7.83sec
	decay on/off	on/off
	decay time	0 – 489.6msec
	EF gain	0 – +60dB

SECTION	PARAMETER	MIN and MAX
EF2 (envelope follower 2 for DCF)	EF source	ch1/ch2/mix/panel key
	EF sidechain LPF freq	16Hz – 20kHz
	EF sidechain HPF freq	16Hz – 20kHz
	threshold level	– 72 – 0dB
	attack time	0 – 489.6msec
	release time	0msec – 7.83sec
	decay on/off	on/off
	decay time	0 – 489.6msec
	EF gain	0 – +60dB
EG (envelope generator)	EG trigger source	ch1/ch2/mix/MIDI note on/panel key
	detector sidechain HPF freq	16Hz – 20kHz
	detector attack time	0 – 489.6msec
	detector release time	0msec – 7.83sec
	trigger mask time	0 – 489.6msec
	threshold level	– 72 – 0dB
	attack time	0msec – 7.83sec
	decay time	0msec – 7.83sec
	sustain level	0 – 100%
	release time	0msec – 7.83sec
LFO	rate	0.025 – 12.8Hz
L.MIDI	[1] – [4] MIDI control number	0 – 31, 64 – 120: control change #0 – #31, #61 – #120/ note on velocity/ channel pressue/ note number (note on)/off
	[1] – [4] MIDI range (min)	0 – 127
	[1] – [4] MIDI range (max)	0 – 127

### Section/Parameter Description

#### EG/trigger mask time

This parameter sets the time from the end of a EG input until the beginning of release of the EG.

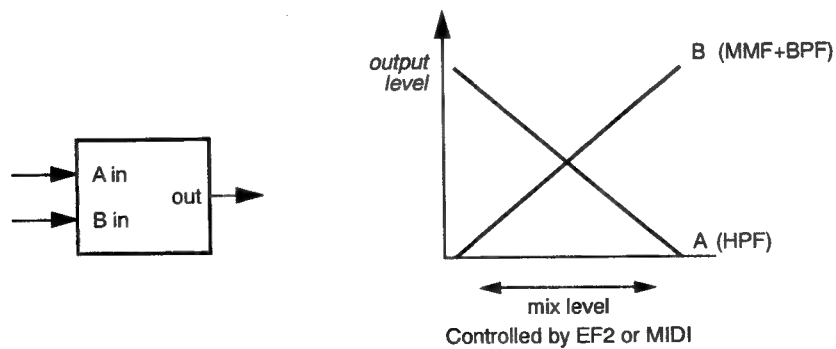
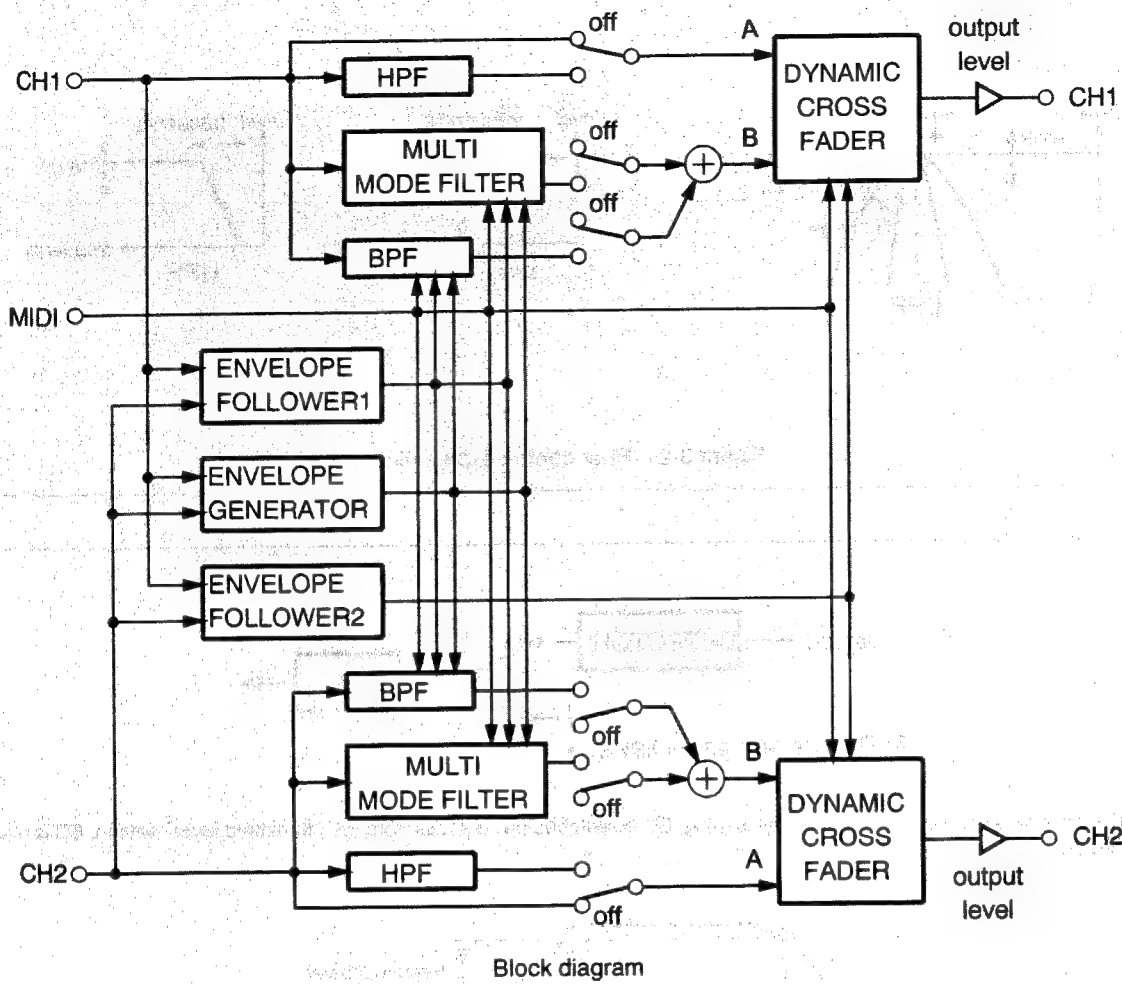


Figure 3-1 – DCF explanation

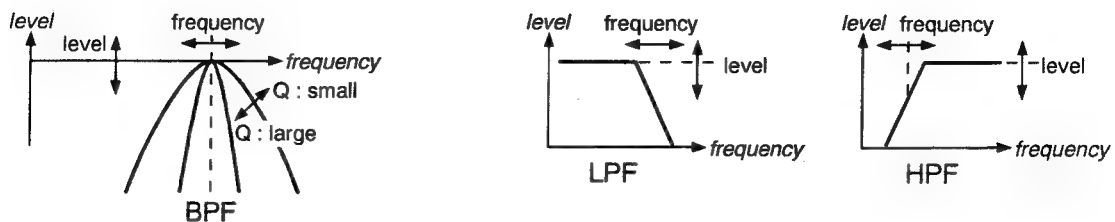
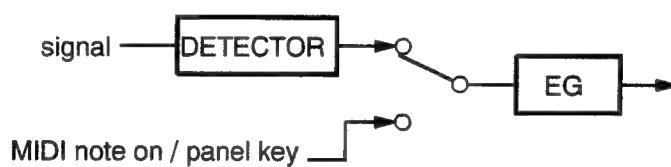


Figure 3-2 – Filter control explanation



\* If the EG input exceeds the "threshold level" setting, EG is switched on . If it falls from the "threshold level" setting, EG is now switched off.

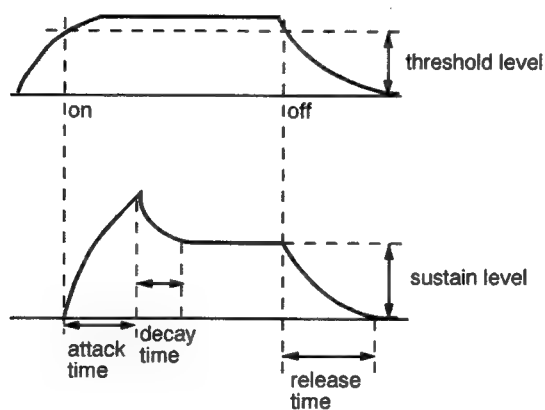


Figure 3-3 – EG explanation

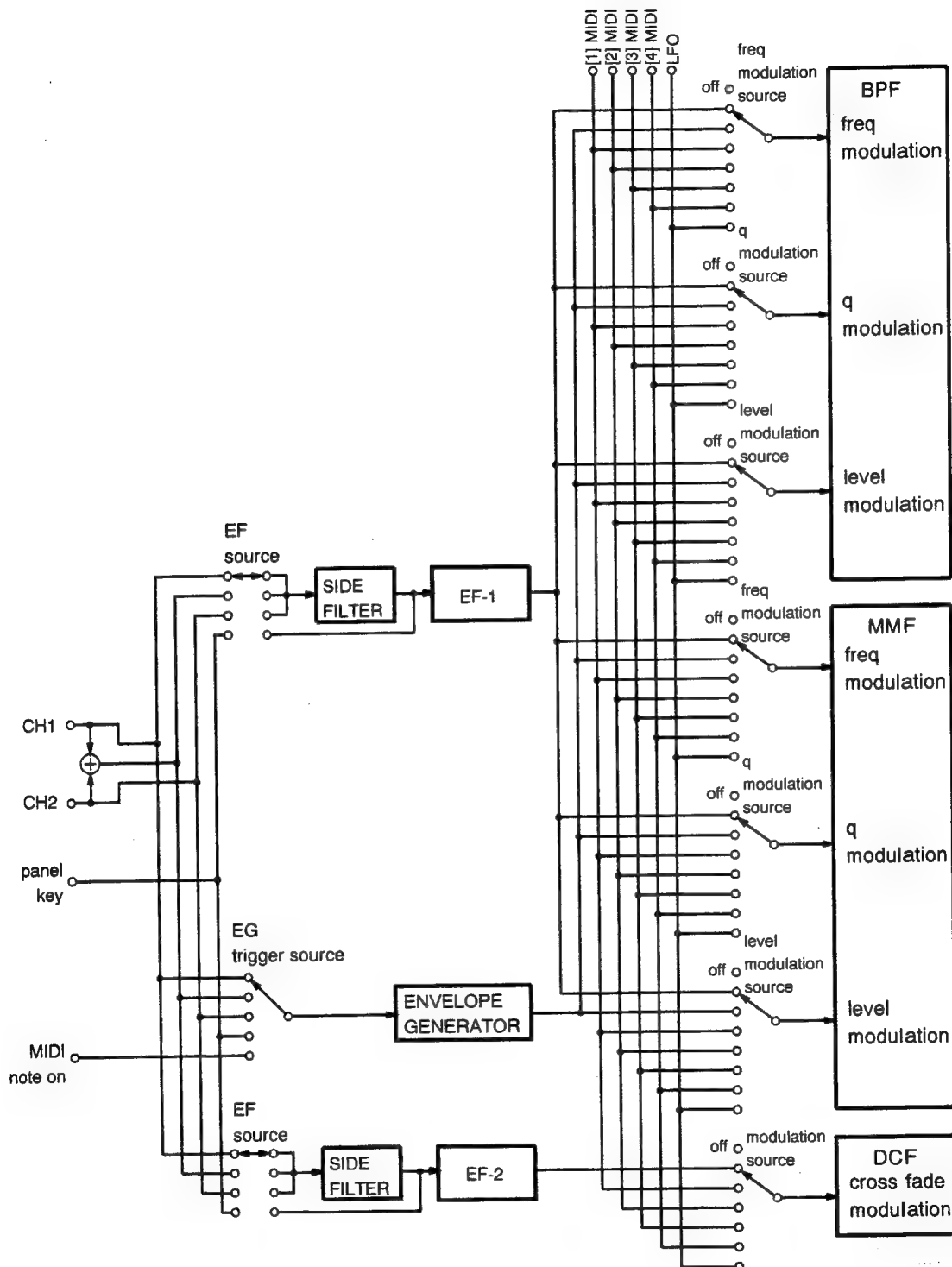


Figure 3-4 – Control system explanation

## Algorithm 4 Extended Compressor Limiter ECL

This is an algorithm based on digital processing compressor (limiter) and expander, which also extends the possibility of these two elements. This algorithm generates two types of gain control signal (A and B), each of which is sent on a different path. This function allows different settings for EQs and sound locations at large amplitude and at small amplitude so that it can be applied to a variety of sounds.

The signal for path A is processed in conventional compressor (limiter) and expander. Then, the signal for path B is the remnants of the entire input signal. Thus, this algorithm is used as a conventional compressor (limiter) and expander if path B is not in use ("path B level" parameter is set 0% and "path B pan (ch1)" parameter is set 0%).

SECTION	PARAMETER	MIN and MAX
DYN (dynamics processor)	basic mode (ch1, ch2)	compressor/ expander
	threshold level (ch1, ch2)	-72 - 0dB
	ratio (ch1, ch2)	1:1 - ∞:1 (compressor), 1:1 - 1:∞ (expander)
	attack time (ch1, ch2)	0 - 489.6msec
	release time (ch1, ch2)	0 - 7.83sec
	side HPF frequency (ch1, ch2)	16Hz - 18kHz
	side PEQ frequency (ch1, ch2)	63Hz - 18kHz
	side PEQ level (ch1, ch2)	-12 - +12dB
	side PEQ q (ch1, ch2)	0.267/0.667/ 1.414/2.145/ 4.319/8.651/ 17.31
	sidein mode	st link on/st link off/ ch2/panel key/MIDI
A.BASS (path A bass)	path A bass on/off	on/off
	path A bass frequency (ch1, ch2)	16Hz - 6.3kHz
	path A bass level (ch1, ch2)	-12 - +12dB
A.TRBL (path A treble)	path A treble on/off	on/off
	path A treble frequency (ch1, ch2)	400Hz - 20.0kHz
	path A treble level (ch1, ch2)	-12 - +12dB
A.PEQ (path A PEQ)	path A PEQ on/off	on/off
	path A PEQ frequency (ch1, ch2)	63Hz - 18kHz
	path A PEQ level (ch1, ch2)	-12 - +12dB
	path A PEQ q (ch1, ch2)	0.267/0.667/ 1.414/2.145/ 4.319/8.651/ 17.31
A.PHS (path A phase offsetter)	path A phase offset on/off	on/off
	path A phase offset mode	1/2/3
B.BASS (path B bass)	path B bass on/off	on/off
	path B bass frequency (ch1, ch2)	16Hz - 6.3kHz
	path B bass level (ch1, ch2)	-12 - +12dB
B.TRBL (path B treble)	path B treble on/off	on/off
	path B treble frequency (ch1, ch2)	400Hz - 20.0kHz
	path B treble level (ch1, ch2)	-12 - +12dB

\* These parameters are valid only when "MIDI" is selected for the "sidein mode" parameter.

SECTION	PARAMETER	MIN and MAX
B.PEQ (path B PEQ)	path B PEQ on/off	on/off
	path B PEQ frequency (ch1, ch2)	63Hz - 18kHz
	path B PEQ level (ch1, ch2)	-12 - +12dB
	path B PEQ q (ch1, ch2)	0.267/0.667/ 1.414/2.145/ 4.319/8.651/ 17.31
B.PHS (path B phase offsetter)	path B phase offset on/off	on/off
	path B phase offset mode	1/2/3
OUTPUT	path A level (ch1, ch2)	0 - 398%
	path A phase (ch1, ch2)	normal/inverse
	path A panpot (ch1, ch2)	0 - 100%
	path B level (ch1, ch2)	0 - 398%
	path B phase (ch1, ch2)	normal/inverse
	path B panpot (ch1, ch2)	0 - 100%
	total level (ch1, ch2)	-18 - 0dB
SCLP (soft clipper)	soft clipper on/off	on/off
	soft clipper level (ch1, ch2)	0/-2/-4/-6dB
L.MIDI	gain recover (ch1, ch2)	on/off
	[1] - [4] control number	0 - 31, 64 - 120: control change #0 - #31, #64 - #120/ note on velocity/ channel pressure/ note number (note on)/off
	[1] - [4] MIDI range (min)	0 - 127
	[1] - [4] MIDI range (max)	0 - 127
	[1] - [4] control parameter	all PEQ parameter/ all bass, treble parameter/ all OUTPUT parameter/ side HPF frequency/ all side PEQ parameter/ all level/all panpot (except on/off parameter) depending on parameter
	[1] - [4] parameter range (min)	depending on parameter
	[1] - [4] parameter range (max)	depending on parameter

### Section/Parameter Description

#### A.PHS, B.PHS (phase offsetter)

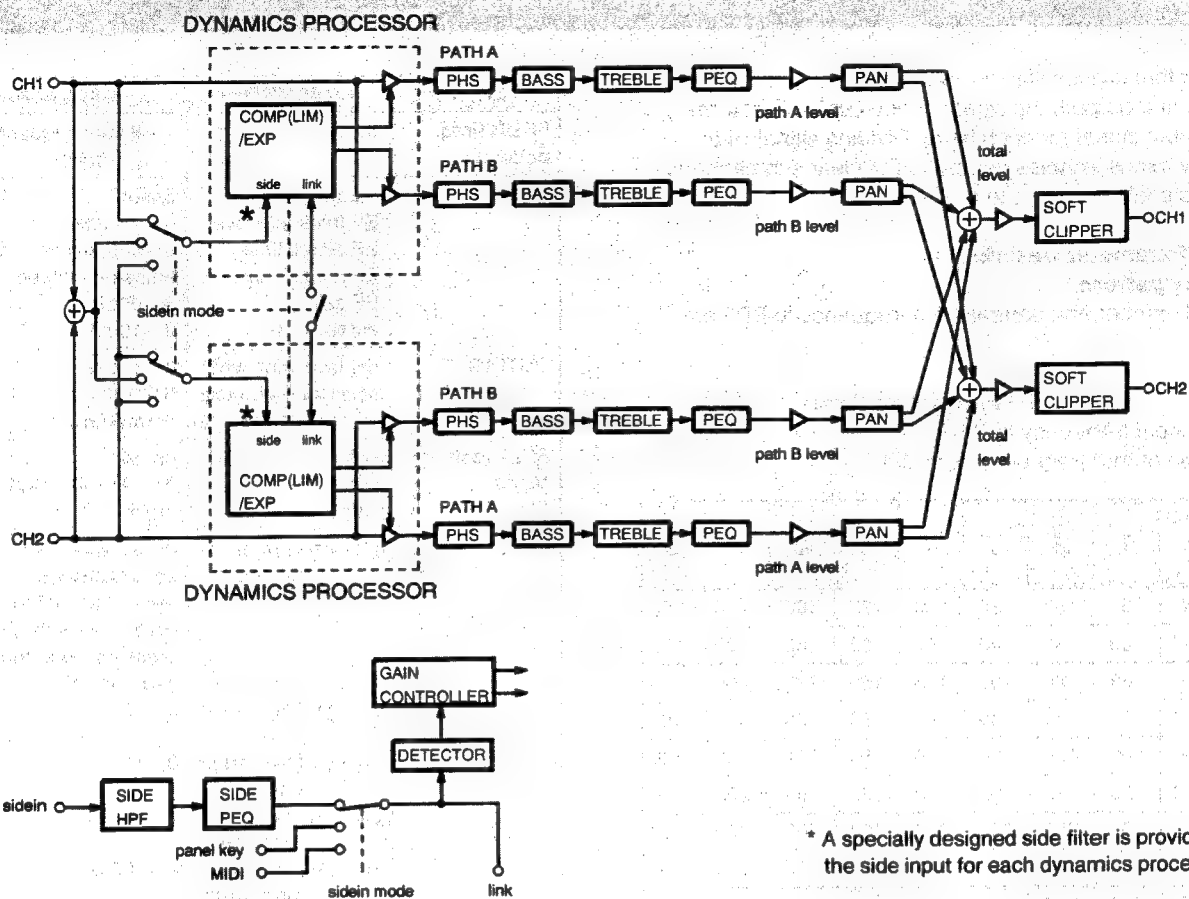
The phase of the middle band is offset to produce an unclear sound location.

#### DYN/sidein mode

- When "st link on" is selected  
The dynamics processor (DYN) works the same for both channels, according to the channel of the greater signal level.
- When "st link off" is selected  
It works independently for each channel.
- When "panel key" is selected  
It works as if the maximum signal level was input each time you press the ENTER button.

#### OUTPUT/path A, B panpot

When the "panpot" parameter is set to "0%", signals input to channel 1 is output through the same channel. (channel 1 to channel 1, channel 2 to channel 2.)



Dynamic processor internal block diagram

Block diagram

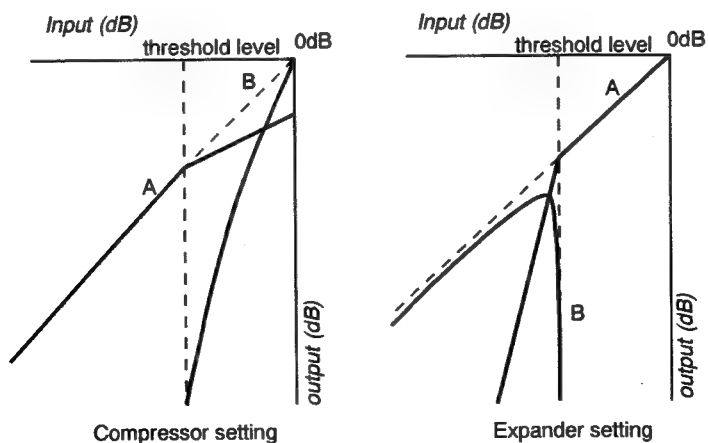


Figure 4-1 – Distribution to paths A and B

0 dB is the maximum input level (see page 12).

## Algorithm 5 Sub-harmonic Generator

SHG

This algorithm divides signals into eight frequency bands, produces and outputs the signal at one octave below the actually input signal for each band. Dividing signals into frequency bands enables generation of clear sub harmonics from the signals you want to process.

### Section/Parameter Description frequency pattern

The band number and central input frequency to BPF are set.

**Table 5-1 Frequency chart**

IN: main input frequency to BPF

OUT: main output frequency from BPF

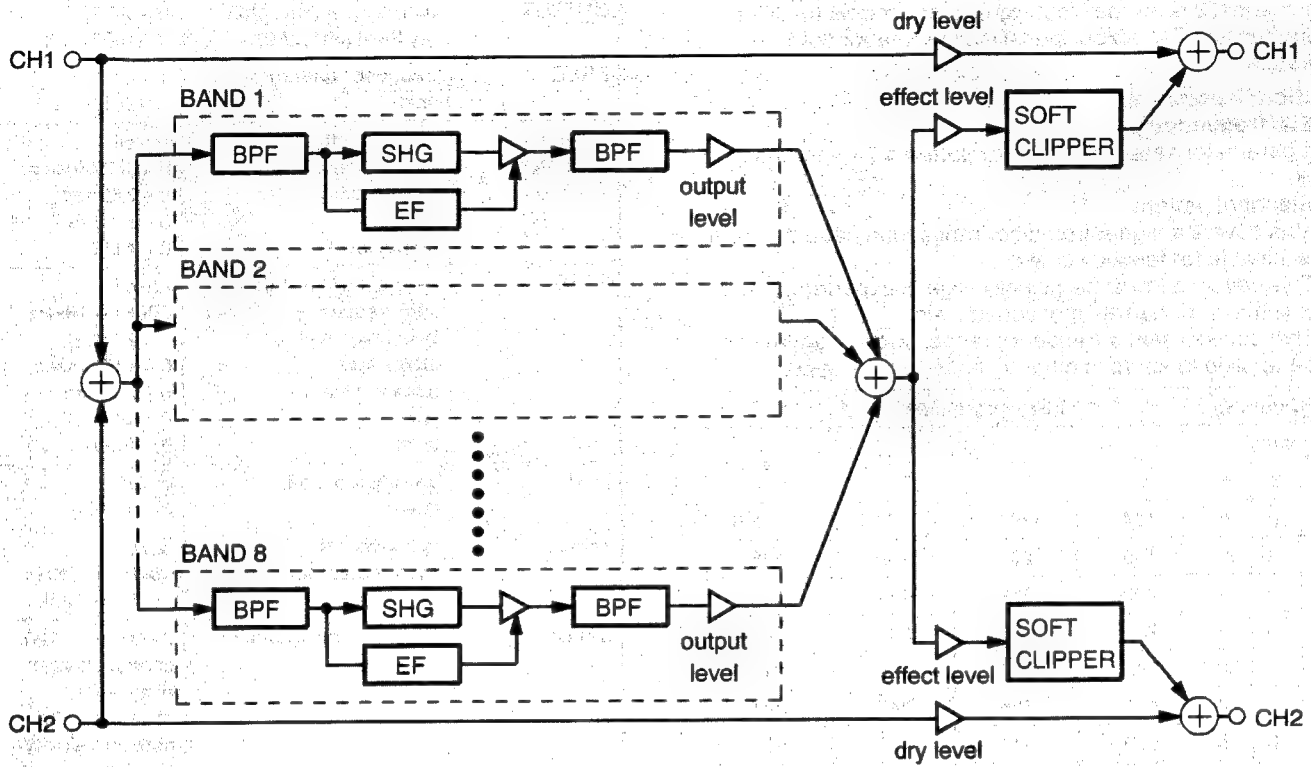
band freq pattern		1	2	3	4	5	6	7	8
1	IN	50	63	80	100	125	160	200	250
	OUT	25	31	40	50	63	80	100	125
2	IN	63	80	100	125	160	200	250	315
	OUT	31	40	50	63	80	100	125	160
3	IN	100	125	160	200	250	315	400	500
	OUT	50	63	80	100	125	160	200	250
4	IN	160	200	250	315	400	500	630	800
	OUT	80	100	125	160	200	250	315	400
5	IN	250	315	400	500	630	800	1000	1250
	OUT	125	160	200	250	315	400	500	630
6	IN	63	90	125	180	250	355	500	710
	OUT	31	45	63	90	125	177	250	355
7	IN	90	125	180	250	355	500	710	1000
	OUT	45	63	90	125	177	250	355	500
8	IN	flat	off	off	off	off	off	off	off
	OUT	flat	off	off	off	off	off	off	off

(Hz)

### Notes

- "frequency patterns 1 – 5" are suitable for any musical instrument.
- "frequency pattern 6" is best suited for the bass guitar, and "frequency pattern 7" is best suited for guitar sounds.
- When "frequency pattern 8" is selected, input signals are not divided by frequency bands and the entire signal is processed through only one unit (one band). The signal output without BPF processing (that is, the signal not divided by frequency bands) includes harmonics.

SECTION	PARAMETER	MIN and MAX
FREQ (freq pattern)	frequency pattern	1 – 8 (see frequency chart)
BAND1 – 8	band on/off	on/off
	EF threshold level	– 72 – 0dB
	EF attack time	0 – 489.6msec
	EF release time	0msec – 7.83sec
	EF gain	0 – 60dB
	output level	0 – 100%
OUTPUT	dry level (ch1, ch2)	0 – 100%
	effect level (ch1, ch2)	0 – 100%
	effect phase (ch1, ch2)	normal/inverse
SCLP (soft clipper)	soft clipper on/off	on/off
	soft clipper level	0/ – 2/ – 4/ – 6dB
	gain recover	on/off
L.MIDI	[1] – [4] control number	0 – 31, 64 – 120: control change # 0 – # 31, # 64 – # 120/ note on velocity/channel pressure/note number (note on)/off
	[1] – [4] MIDI range (min)	0 – 127
	[1] – [4] MIDI range (max)	0 – 127
	[1] – [4] control parameter	effect level
	[1] – [4] parameter range (min)	0 – 100%
	[1] – [4] parameter range (max)	0 – 100%



Block diagram

## Algorithm 6 Channel Vocoder VOC

This is a vocoder controlled by frequency band division. Channel 1 is for the input signal that passes the filter bank and channel 2 is for the input signal that controls the filter characteristic. The output signal is the same for both channels.

### Section/Parameter Description

#### FREQ/frequency pattern

This parameter sets the frequency patterns 1 – 4 for the filter bank.

1. Standard pattern
2. This covers a higher frequency range than pattern 1, so it is suitable for female vocals.
3. This covers a lower frequency range than pattern 1, so it is suitable for narration or speech, etc.
4. This covers a wider frequency range, so this pattern can be applied to sources other than the human voice.

band number			frequency pattern (Hz)			
shift			1	2	3	4
–	0	+				
1	*	*	124	207	97.4	93.6
2	1	*	165	261	130	125
3	2	1	220	319	184	167
4	3	2	277	391	245	223
5	4	3	340	479	292	297
6	5	4	416	586	347	397
7	6	5	509	717	413	520
8	7	6	623	878	491	707
9	8	7	763	1.07K	584	944
10	9	8	933	1.32K	694	1.26K
11	10	9	1.14K	1.61K	825	1.68K
12	11	10	1.40K	1.97K	982	2.24K
13	12	11	1.71K	2.41K	1.17K	3.00K
14	13	12	2.10K	2.95K	1.39K	4.00K
15	14	13	2.80K	3.72K	1.85K	5.34K
16	15	14	3.73K	4.97K	2.47K	7.13K
*	16	15	4.98K	6.63K	3.30K	9.51K
*	*	16	6.65K	8.85K	4.41K	12.70K

The “shift” parameter allows frequency shift for channel 1. When “0” is selected for the “shift” parameter, frequency settings for both channel become the same. See Figure 6-1.

#### PASS

This section controls the process through which the input signal to channel 2 is output. This is for consonant sounds and is not equipped with a HPF and gate.

#### PASS/pass control on/off

When this parameter is set to “off,” no signal comes out from the PASS CONTROL (see Block diagram).

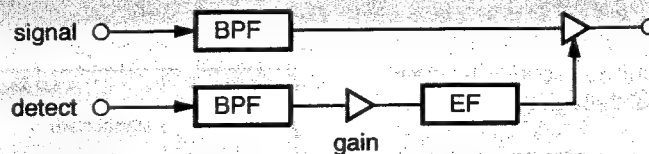
#### BAND1 – 16/band on/off

When this parameter is set to “off,” no signal comes out from any BAND (see Block diagram).

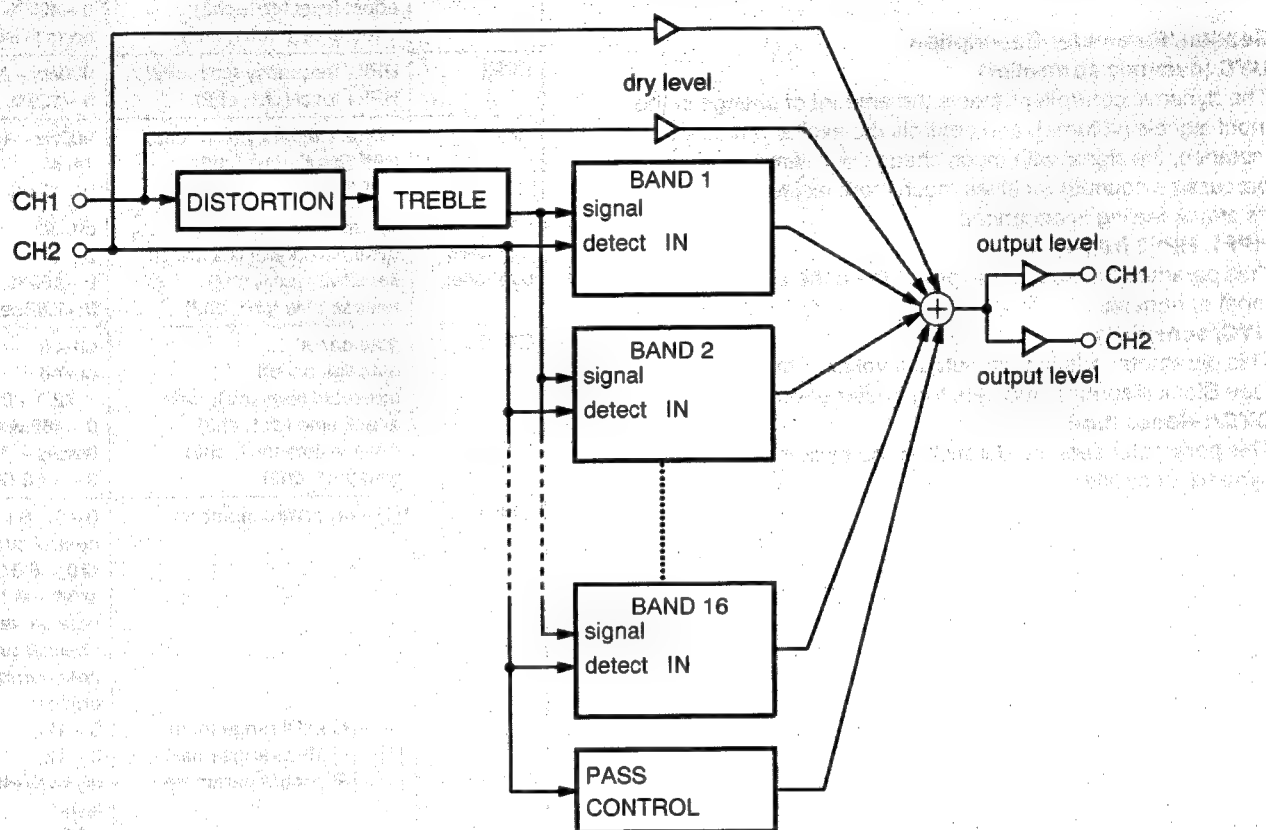
#### DIST/distortion

The input signal of channel 1 is distorted and the harmonic overtone is added.

SECTION	PARAMETER	MIN and MAX
OUTPUT	output level (ch1, ch2)	0 – 100%
	dry level (ch1, ch2)	0 – 100%
FREQ	frequency pattern shift	1 – 4 – /0/ +
BAND1-BAND16	band on/off	on/off
	attack time	0 – 489.6msec
	release time	0 – 7.83sec
	gain	0 – +12dB
	output level	0 – 400%
PASS	pass control on/off	on/off
	HPF frequency	500Hz – 18kHz
	threshold level	–72 – 0dB
	attack time	0 – 489.6msec
	release time	0 – 7.83sec
	gain level	0 – +36dB 0 – 100%
DIST	distortion on/off drive	on/off 0 – +36dB
TRBL	treble on/off treble frequency treble level	on/off 400Hz – 20KHz –12 – +12dB
L.MIDI	[1] – [4] control number	0 – 31, 64 – 120: control change #0 – #31, #64 – #120/ note on velocity/ channel pressure/ note number (note on)/off
	[1] – [4] MIDI range (min)	0 – 127
	[1] – [4] MIDI range (max)	0 – 127
	[1] – [4] control parameter	output level/ dry level/drive/ treble frequency/ treble level
	[1] – [4] parameter range (min)	depending on parameter
	[1] – [4] parameter range (max)	depending on parameter



BAND internal block diagram



Block diagram

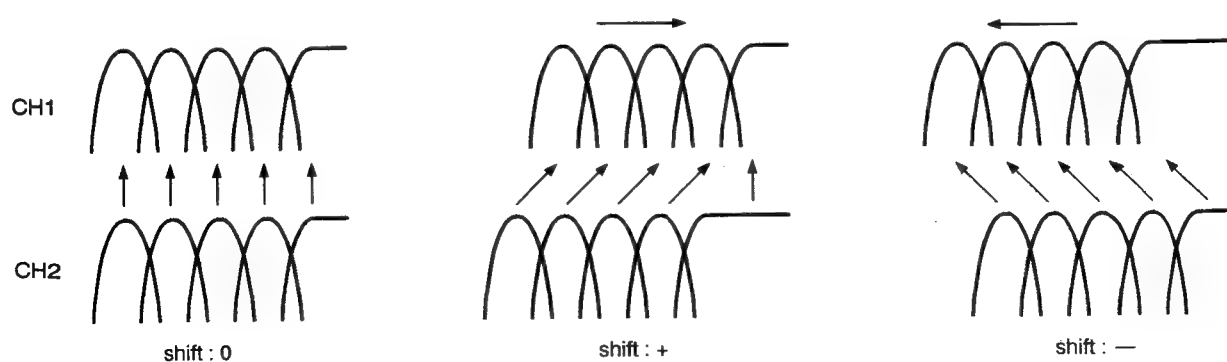


Figure 6-1 - Shift explanation

## Algorithm 7 Exciter EXC

This algorithm produces a sharply outlined sound with clear attack by mixing the high band signal through the HPF and the delayed original sound.

There are two types of HPF: HPF1 and HPF2. The former enhances the outline of sound, and the latter enhances the high frequency band.

### Section/Parameter Description

#### DYC (dynamic controller)

The dynamic controller detects the amount of change in the input signals (volume), and controls the exciter effect. For instance, the signal with much change in volume (such as percussion sounds) receives much more exciter effect, and its attack feeling is enhanced.

#### HPF1, HPF2 frequency

This parameter indicates the point where the exciter effect is most enhanced.

#### DYC/sensitivity

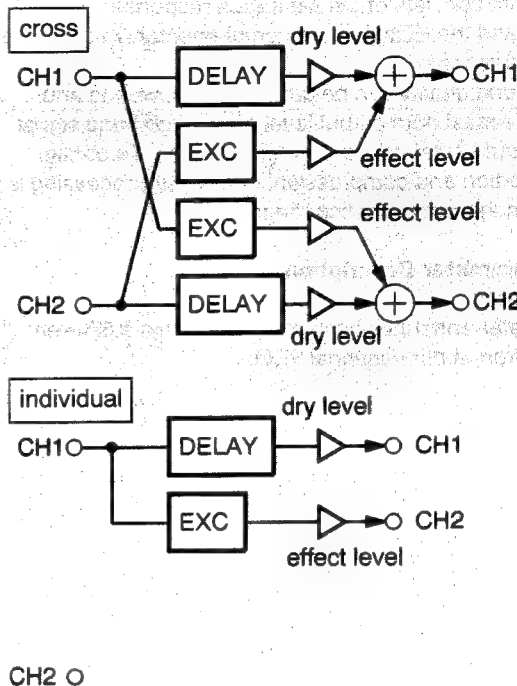
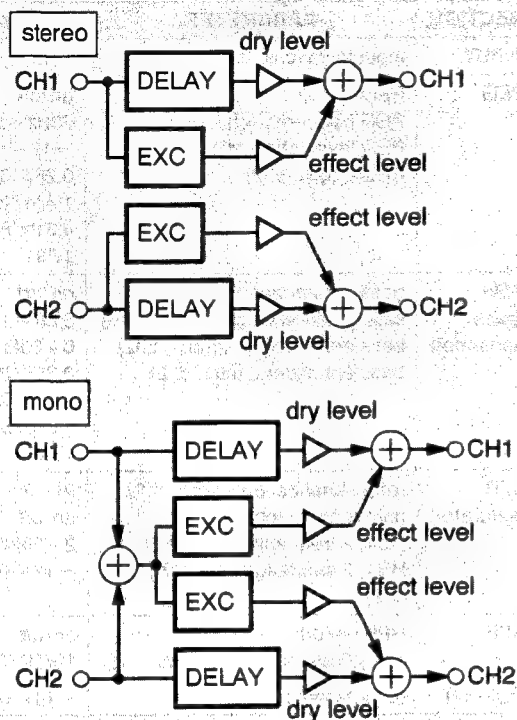
This parameter adjusts the output level from the DETECTOR (see Block diagram), and sets the exciter effect.

#### DYC/release time

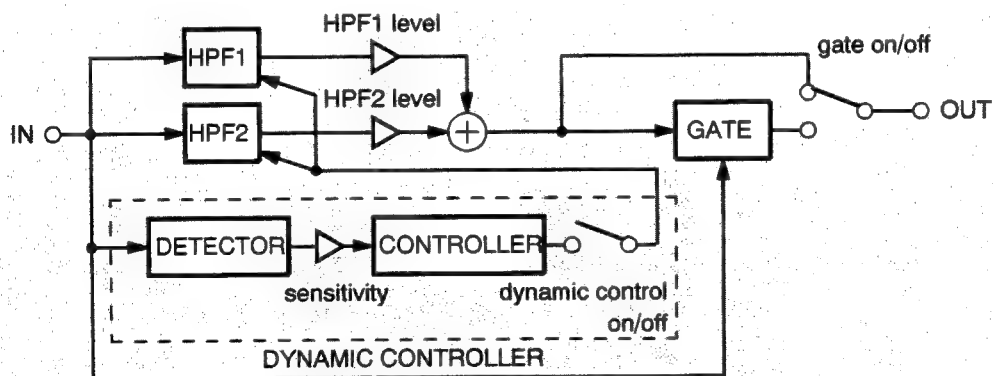
This parameter sets the duration of the exciter effect by the dynamic controller.

SECTION	PARAMETER	MIN and MAX
PATH	signal path	stereo/cross/ individual/mono
OUTPUT	dry level (ch1, ch2) dry signal delay (ch1, ch2) effect level (ch1, ch2) effect phase (ch1, ch2)	0 - 100% 0 - 0.458msec 0 - 400% normal/inverse
HPF1	HPF1 frequency (ch1, ch2) HPF1 level (ch1, ch2)	500Hz - 24.0kHz 0 - 100%
HPF2	HPF2 frequency (ch1, ch2) HPF2 peak (ch1, ch2) HPF2 level (ch1, ch2)	500Hz - 18.0kHz 1 - 4 0 - 100%
DYC (dynamic controller)	dynamic control on/off dynamic control link on/off sensitivity (ch1, ch2) release time (ch1, ch2)	on/off on/off 0 - 100% 0 - 7.83sec
GATE	gate on/off gate link on/off threshold level (ch1, ch2) attack time (ch1, ch2) release time (ch1, ch2) gain (ch1, ch2)	on/off on/off - 72.0 - 0.0dB 0 - 489.6msec 0msec - 7.83sec 0 - + 60.0dB
L.MIDI	[1] - [4] control number  [1] - [4] MIDI range (min) [1] - [4] MIDI range (max) [1] - [4] control parameter  [1] - [4] parameter range (min) [1] - [4] parameter range (max)	0 - 31, 64 - 120: control change # 0 - # 31, # 64 - # 120/ note on velocity/ channel pressure/ note number (note on)/off 0 - 127 0 - 127 dry level/effect level/ HPF1 frequency/ HPF1 level/ HPF2 frequency/ HPF2 peak/ HPF2 level/ sensitivity depending on parameter depending on parameter

signal path



EXC internal block diagram



Block diagram

## Algorithm 8

## Non-linear Saturator

NLS

This algorithm consists of the saturation response processing and the EQs that are complementarily set before and after the processing.

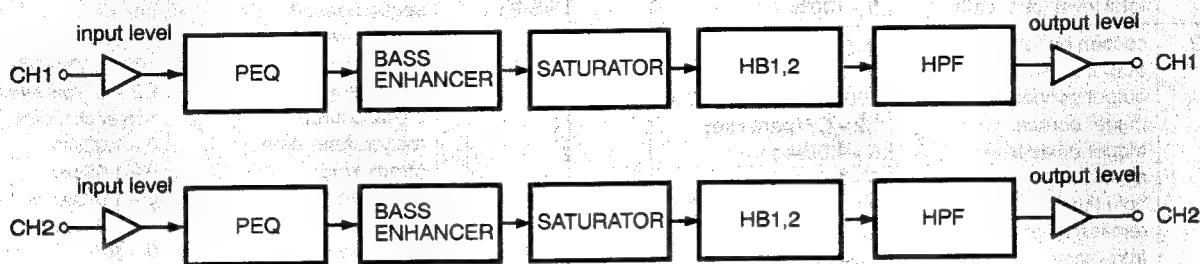
A unique sound quality can be produced by lowering and distorting the maximum output level of the high band signal when the input signal level is high (Figure 8-1). To obtain natural distortion and compression, non-linear processing is performed in the oversampling range.

### Section/Parameter Description

#### bass enhancer

This parameter enhances bass. You can obtain a different effect than from a conventional PEQ.

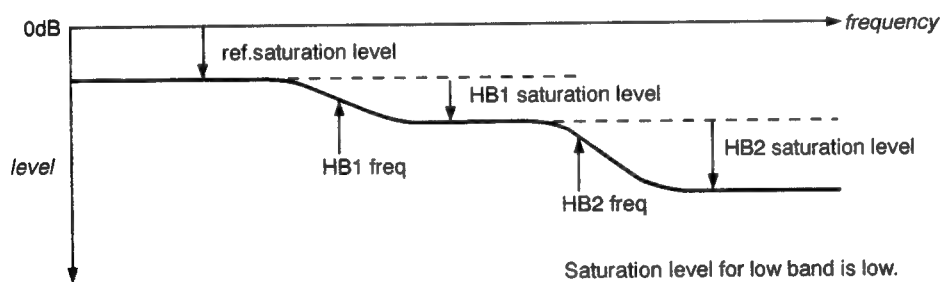
SECTION	PARAMETER	MIN and MAX
INPUT	input level (ch1, ch2)	-12 - +12dB
PEQ	PEQ on/off PEQ frequency (ch1, ch2) PEQ level (ch1, ch2) PEQ q (ch1, ch2)	on/off 63Hz - 20kHz -12 - +12dB 0.267/0.667/ 1.414/2.145/ 4.319/8.651/ 17.31
BEH (bass enhancer)	bass enhancer on/off bass enhancer frequency (ch1, ch2) bass enhancer level (ch1, ch2) bass enhancer q (ch1, ch2)	on/off 31.5 - 125Hz 0 - 6dB 0.267/0.667/ 1.414/2.145/ 4.319/8.651/ 17.31
STR (saturator)	ref. saturation level (ch1, ch2) gain recover (ch1, ch2) HB1, 2 frequency (ch1, ch2) HB1, 2 saturation level (ch1, ch2)	0/-3/-6dB on/off 2 - 20kHz -12 - 0dB
HPF	HPF on/off cutoff frequency (ch1, ch2)	on/off 10/20/30Hz
OUTPUT	output level (ch1, ch2)	-18 - 0dB



Block diagram

Saturation level is determined by the "ref. saturation level" and "HB (high band) 1, 2 saturation level" parameter; therefore, the saturation level for high band is low. The "ref. saturation level" parameter is effective for all frequency bands, and the hard clip characteristic functions when it is set to "0 dB."

The function of the "gain recover" parameter is the same as explained for the "soft clipper" parameter on page 14.



0 dB is the maximum digital level.

Figure 8-1 – Maximum output characteristics

## Algorithm 9 Percussion Synthesizer

PRC

A solid percussion sound is synthesized by such a method as used in an old-fashioned rhythm machine or analog synthesizer. This consists of a total of 20 sound generators, categorized into 7 types.

SECTION	PARAMETER	MIN and MAX
OUTPUT	total level (ch1, ch2)	0 - 100%
SIN-A1, 2	section on/off output level output panpot trigger source trigger delay time attack time hold time release time frequency fine sweep intensity sweep time drive velocity type	on/off 0 - 200 100%:0% - 0%:100% C2 - C7/panel key 0 - 1.02sec 0 - 1.02sec 0 - 1.02sec 0 - 7.94sec C0 - G9 -50 - +50cent -4800 - +4800cent 0 - 7.94sec off/1/2 off/1/2/3
SIN-B1 - 4	section on/off output level output panpot trigger source trigger delay time attack time hold time release time frequency fine sweep intensity sweep time drive velocity type	on/off 0 - 200 100%:0% - 0%:100% C2 - C7/panel key 0 - 1.02sec 0 - 1.02sec 0 - 1.02sec 0 - 7.94sec C0 - G9 -50 - +50cent -4800 - +4800cent 0 - 7.94sec off/1/2 off/1/2/3
SIN-C1 - 3	section on/off output level output panpot trigger source trigger delay time attack time hold time release time frequency A fine A frequency B fine B HPF frequency output A level output B level output R level velocity type	on/off 0 - 200 100%:0% - 0%:100% C2 - C7/panel key 0 - 1.02sec 0 - 1.02sec 0 - 1.02sec 0 - 7.94sec C0 - G9 -50 - +50cent -50 - +50cent 31.5Hz - 20kHz 0 - 100 0 - 100 0 - 100 off/1/2/3
NS-A1 - 7	section on/off output level output panpot trigger source trigger delay time attack time hold time release time dry level BPF frequency BPF level BPF q velocity type	on/off 0 - 100 100%:0% - 0%:100% C2 - C7/panel key 0 - 1.02sec 0 - 1.02sec 0 - 1.02sec 0 - 7.94sec 0 - 100 63Hz - 18kHz -100 - 100 0.267/0.667/1.414/ 2.145/4.319/8.651/17.31 off/1/2/3

SECTION	PARAMETER	MIN and MAX
NS-B1	section on/off output level output panpot trigger source signal source trigger delay time attack time hold time release time BPF frequency BPF q BPF sweep intensity BPF sweep time velocity type	on/off 0 - 100 100%:0% - 0%:100% C2 - C7/panel key noise/ch1/ch2 0 - 1.02sec 0 - 1.02sec 0 - 1.02sec 0 - 7.94sec 0 - 100 0 - 100 -100 - +100 0 - 1.02sec off/1/2/3
NS-C1	section on/off output level output panpot trigger source trigger delay time noise1 release time noise1 repeat mode noise1 repeat interval noise1 level noise2 release time noise2 offset time noise2 level bass frequency bass level treble frequency treble level PEQ frequency PEQ level PEQ q velocity type	on/off 0 - 100 100%:0% - 0%:100% C2 - C7/panel key 0 - 1.02sec 0 - 489.6msec 1/2 0 - 101msec 0 - 100 0 - 7.83sec 0 - 101msec 0 - 100 16Hz - 6.3kHz -12 - +12dB 400Hz - 20kHz -12 - +12dB 63Hz - 20kHz -12 - +12dB 0.267/0.667/1.414/ 2.145/4.319/8.651/ 17.31 off/1/2/3
PLS-A1, 2	section on/off output level output panpot trigger source trigger delay time tap position feedback pulse width velocity type	on/off 0 - 100 100%:0% - 0%:100% C2 - C7/panel key 0 - 1.02sec 0 - 100 0 - 100 0 - 101msec off/1/2/3
L.MIDI	[1] - [4] control number  [1] - [4] MIDI range (min) [1] - [4] MIDI range (max) [1] - [4] control parameter  [1] - [4] parameter range (min) [1] - [4] parameter range (max)	0 - 31, 64 - 120: control change #0 - #31, #64 - #120/ channel pressure/off 0 - 127 0 - 127 all level/all output panpot/all frequency/ all drive/all q depending on parameter depending on parameter

**Section/Parameter Description**

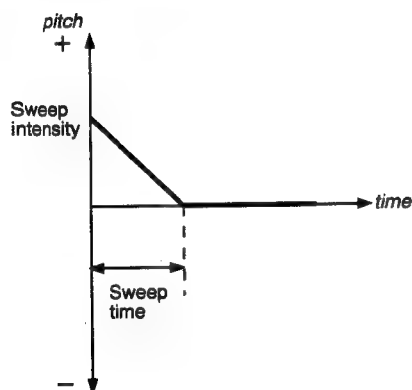
SIN-A1,2	: Sine wave oscillator
SIN-B1 - 4	: Sine wave oscillator
SIN-C1 - 3	: Sine wave oscillator + Ring modulator + EQ
NS-A1 - 7	: Noise generator + BPF
NS-B1	: Noise generator + BPF with sweep function
NS-C1	: Multi-tap noise generator
PLS-A1,A2	: Pulse generator + resonator

**SIN-A1, A2**

This section enables polyphonic operation by generating sounds by two triggers alternately.

**SIN-A1,2, SIN-B1 - 4/sweep, NS-B1/BPF sweep**

The "sweep intensity" parameter sets the pitch at the beginning of sweep.

**NA-A1 - 7**

This section sets the tone by the "BPF level" and "dry level" parameters.



BPF    BPF+dry    BPF level = -25, dry level = 100

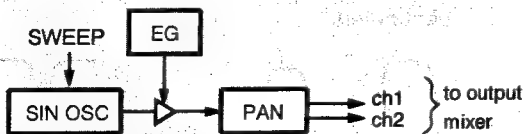
**NS-C1**

This section consists of two groups: noise 1 and noise 2. Noise 1, then, consists of several noise pulses. The interval among them is set by the "repeat interval" parameter and their repeat pattern is set by the "repeat mode" parameter.

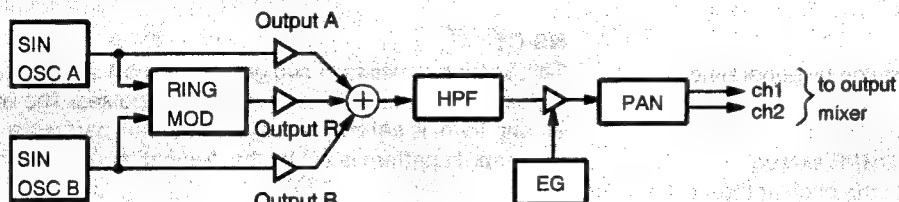
**PLS-A1,2**

The tone and length of sound varies according to the settings of the "tap position" and "feedback" parameters.

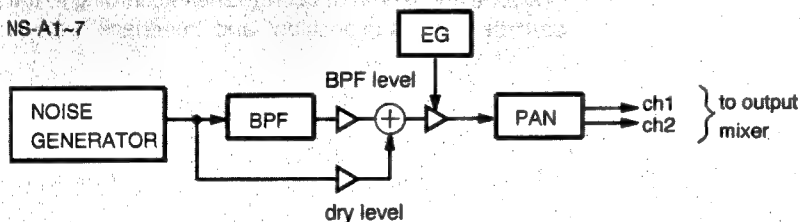
SIN-A1,2 B1-4



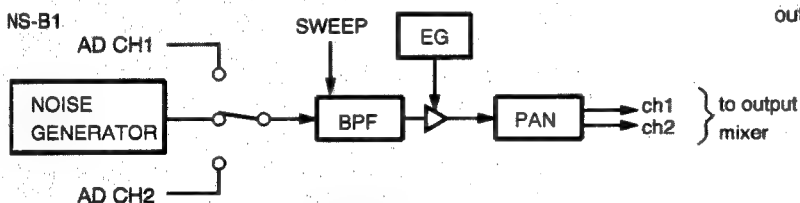
SIN-C1-3



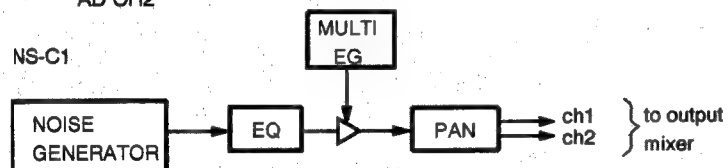
NS-A1-7



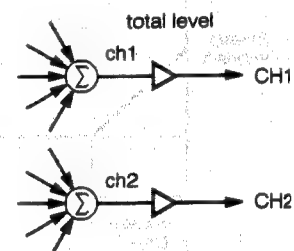
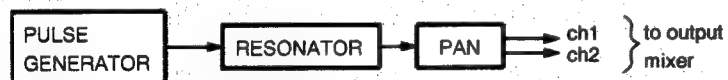
NS-B1



NS-C1



PLS-A1,2



Block diagram

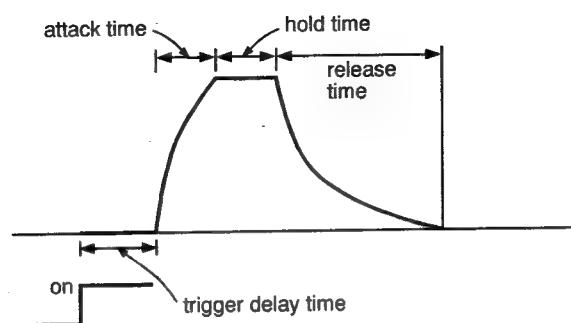


Figure 9-1 – EG explanation

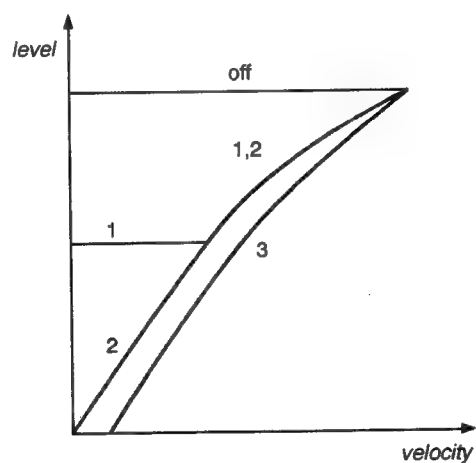


Figure 9-2 – Velocity type (off/1/2/3).

## Algorithm 10 Monophonic Synthesizer SYN

This algorithm is a powerful monophonic synthesizer composed of two analog-like OSCs (oscillators), two newly developed OSCs with a noise generator and a comb filter, one LPF, two BPFs, one HPF, and six EGs.

SECTION	PARAMETER	MIN and MAX
MODE	trigger mode	single 1/single 2/ single 3/ multi 1/multi 2 except 1/except 2
	portamento mode	off/full
	portamento time	time/fingered 0 - 5.12sec
OSC1, 2	OSC on/off	on/off
	octave	32/16/8/4/2
	coarse	0 - 11
	fine	-50 - +50cent
	waveform basic	saw/square/triangle
	waveform edge	hard/soft
	output level	0 - 100
	output EG select	thru/EG1 - EG4
	output path	path 1/path 2
	shape factor	0 - 100
	wave shape mod source	off/EG5/EG6/LFO
	wave shape mod intensity	-100 - +100
OSC3, 4	OSC on/off	on/off
	octave	32/16/8/4/2
	coarse	0 - 11
	fine	-50 - +50cent
	output level	0 - 100
	output EG select	thru/EG1 - EG4
	output path	path 1/path 2
	initial resonance	0 - 100
	resonance mod source	off/EG5/EG6
	resonance mod intensity	-10 - +10
P.EG (pitch EG)	start pitch	-4800-+4800cent
	break pitch	-4800-+4800cent
	time 1	0 - 15.87sec
	time 2	0 - 15.87sec
W.LFO	rate	0 - 100
LPF	LPF on/off	on/off
	input	path 1/path 2
	frequency	0 - 100
	output level	0 - 100
	output EG select	EG1/EG2
	KBD track break key	C1 - C8
	KBD track intensity	-200 - +200
	filter EG select	off/EG5/EG6
	filter EG intensity	-100 - +100

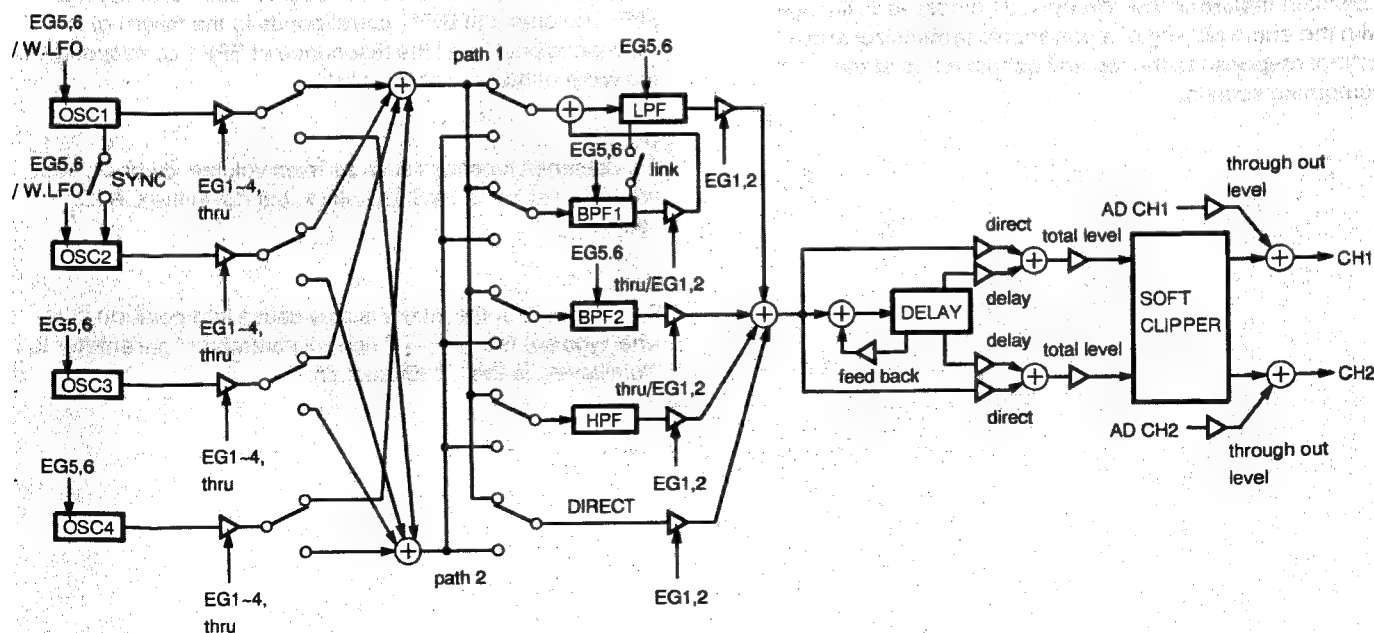
SECTION	PARAMETER	MIN and MAX
BPF1, 2	BPF1, 2 on/off	on/off
	input	path 1/path 2
	frequency	0 - 100
	resonance	0 - 100
	output level	0 - 100
	output EG select	thru*2/EG1/EG2
	KBD track break key	C1 - C8
	KBD track intensity	-200 ~ +200
	filter EG select	off/EG5/EG6
	filter EG intensity	-100 - 100
HPF	LPF-BPF1 link on/off	on/off
	HPF on/off	on/off
	input	path 1/path 2
	frequency	0 - 100
	output level	0 - 100
DIRECT	output EG select	EG1/EG2
	direct on/off	on/off
	input	path 1/path 2
OUTPUT	output level	0 - 100
	output EG select	EG1/EG2
	total level	0 - 100%
	direct level (ch1, ch2)	0 - 100%
	delay time (ch1, ch2)	0 - 1365.31msec
	delay level (ch1, ch2)	0 - 100%
	delay phase (ch1, ch2)	normal/inverse
	feedback time	0 - 1365.31msec
	feedback level	0 - 99%
	feedback phase	normal/inverse
SCLP (soft clipper)	through out level	0 - 100%
	soft clipper on/off	on/off
	soft clipper level	0/-2/-4/-6dB
EG1 - 4	gain recover	on/off
	trigger delay time	0 - 15.87sec
	attack time	0 - 7.83sec
	decay time	0 - 7.83sec
	sustain level	0 - 100%
	release time	0 - 7.83sec
	KBD track break key	C1 - C8
	level KBD track	-L100 - -R100
	time KBD track	0 - 100
	time KBD track SW (attack)	- /off/ +
	time KBD track SW (decay)	- /off/ +
	time KBD track SW (release)	- /off/ +
	velocity level intensity	0 - 100
	velocity time intensity	0 - 100
	velocity time SW (attack)	- /off/ +
	velocity time SW (decay)	- /off/ +
	velocity time SW (release)	- /off/ +

\*1: "effect from OSC1" is selectable only in the OSC2 section.

\*2: "thru" is only for BPF1.

SECTION	PARAMETER	MIN and MAX
EG5,6	trigger delay time	0 – 15.87sec
	attack time	0 – 15.87sec
	decay time	0 – 15.87sec
	sustain level	0 – 100%
	release time	0 – 15.87sec
	KBD track break key	C1 – C8
	level KBD track	– L100 – – R100
	time KBD track	0 – 100
	time KBD track SW (attack)	– /off/ +
	time KBD track SW (decay)	– /off/ +
	time KBD track SW (release)	– /off/ +
	velocity level intensity	0 – 100
	velocity time intensity	0 – 100
	velocity time SW (attack)	– /off/ +
	velocity time SW (decay)	– /off/ +
	velocity time SW (release)	– /off/ +
BEND (pitch bend)	OSC1 – 4 pitch bend on/off	on/off
	filter pitch bend on/off	on/off
	pitch bend range	0 – 1200cent
VIB (vibrato)	vibrato rate	0.1 – 12.8Hz
	vibrato intensity control	0 – 31, 64 – 120: control change #0 – #31, #64 – #120/ channel pressure/off
	OSC1 – 4 vibrato on/off	on/off
	OSC1 – 4 initial intensity	0 – 100
	OSC1 – 4 max intensity	0 – 100

SECTION	PARAMETER	MIN and MAX
L.MIDI	[1] – [4] control number	0 – 31, 64 – 120: control change #0 – #31, #64 – #120/ channel pressure/off
	[1] – [4] MIDI range (min)	0 – 127
	[1] – [4] MIDI range (max)	0 – 127
	[1] – [4] control parameter	OSC1 – 4 output level/OSC3,4 initial resonance/LPF frequency/LPF output level/BPF1 frequency/ BPF1 output level/ BPF2 frequency/ BPF2 output level/ HPF frequency/ HPF output level/ DIRECT output level/ total level/ feedback level/ direct level/delay level/ through out level depending on parameter
	[1] – [4] parameter range (min)	depending on parameter
	[1] – [4] parameter range (max)	depending on parameter



Block diagram

## Section/Parameter Description

### trigger mode

The processing that assigns the note data (data from pressed keys) to a limited number of sound generation units is generally called *key assignment*. The EG operation setting for the multiple note data of single-tone instruments is called *trigger mode*. The "trigger mode" parameter sets these two elements.

The "single1," "single2" and "single3" of the "trigger mode" parameter operate the EG by a single trigger.

During note on in the single trigger mode, the EG is not influenced by the latter note on (another note-on during the former note-on). (See Figure 10-1)

The priority given to the interval varies for the "single1," "single2" and "single3" settings.

single1:	Last note priority
single2:	High note priority
single3:	Low note priority

In either the "multi1" and "multi2" modes, the interval is assigned by latter tone priority and the trigger is a multi-trigger. In multi-trigger mode, the EG starts every time note-on data is received. Both "multi1" and "multi2" are multi-trigger mode but they function differently. (See Figure 10-1.) "except1" and "except2" modes are special key assignment modes designed for the linked operation with a polyphonic synthesizer in an actual live concert. The trigger is a multi-trigger. In "except1" mode, the interval is assigned by high tone priority and in "except2" mode it is assigned by low tone priority. The difference of the "except1, 2" modes from other modes is that, when a generated note becomes off, the next note with priority is not generated, and the EG is set to release mode instead.

The main feature of the "except1, 2" modes is its linkage with the chord playing of a polyphonic synthesizer and a natural response to the top and bottom notes of the composite sounds.

### portamento mode

**full time:** The portamento is always applied.

**fingered:** The portamento is applied only when legato playing is performed.

### OSC2/effect from OSC1

Aliasing noise is generated because the interpolation operation is not executed for the "sync" and "mod" operations.

### OSC3, 4

These are new types of sound sources, each made up of a noise generator and a comb filter. When the "initial resonance" parameter is set to "0," they function as noise generators. As the value of this parameter increases, the pitch feeling (mixture ratio of noise : interval) become clearer.

The pitch can be modulated in real time by using either the EG5 or EG6 section, which you can select with the "resonance mod source" parameter. At the same time, you can set the modulation depth by using the "resonance mod intensity" parameter.

### W. LFO

This is the LFO that modulates the wave shapes of OSC1 and OSC2.

### LPF-BPF1 link

When this parameter is set to "on," the values of other parameters, except the "input path," "resonance," "output EG select" and "output level" parameters, are ignored and the same values selected in the LPF section are used instead.

When the values of the other parameters are set to the same values as set in the LPF section, with this parameter "on," BPF1 can be used at almost the same resonance level of LPF. The output of BPF1 corresponds to the height of the resonance peak, and the resonance of BPF1 corresponds to the width of the resonance peak.

### On MIDI

Fundamental functions such as main volume, dumper, etc. are to be set in the SYS. SG block, but not in the L.MIDI section.

### Note

Rapid control of the intervals may cause interpolation noise when you set the "[1] - [4] control parameter" parameter to "total level" in the L.MIDI section.

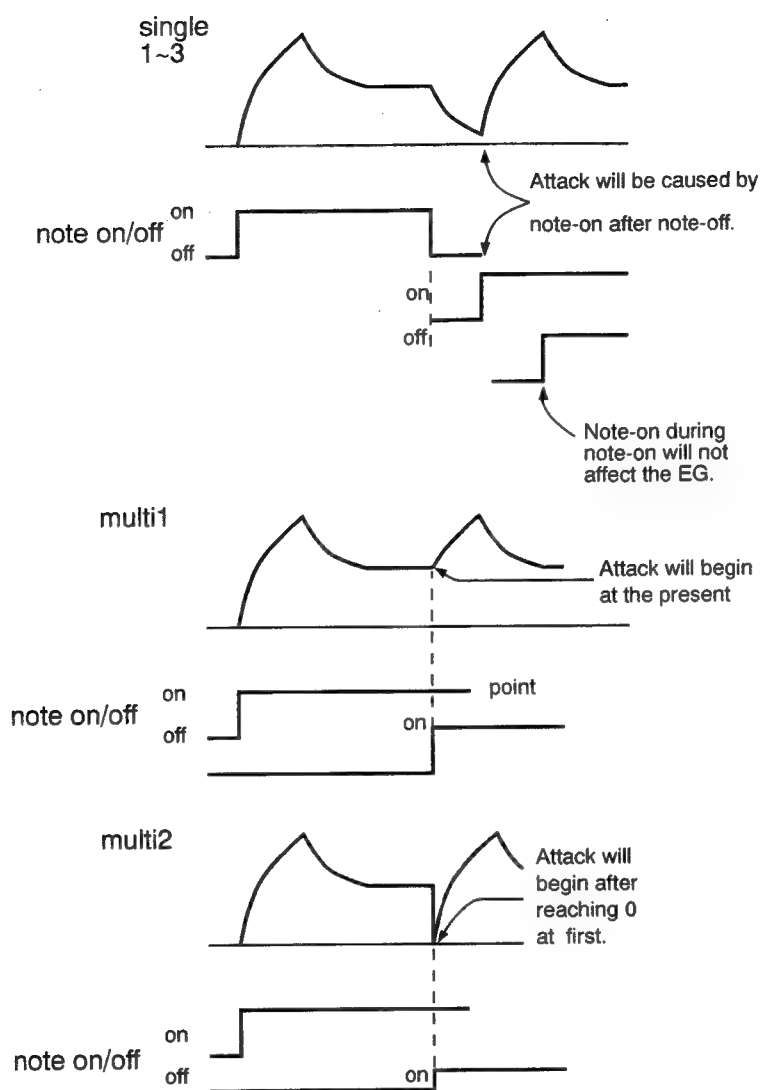


Figure 10-1 – Key mode explanation

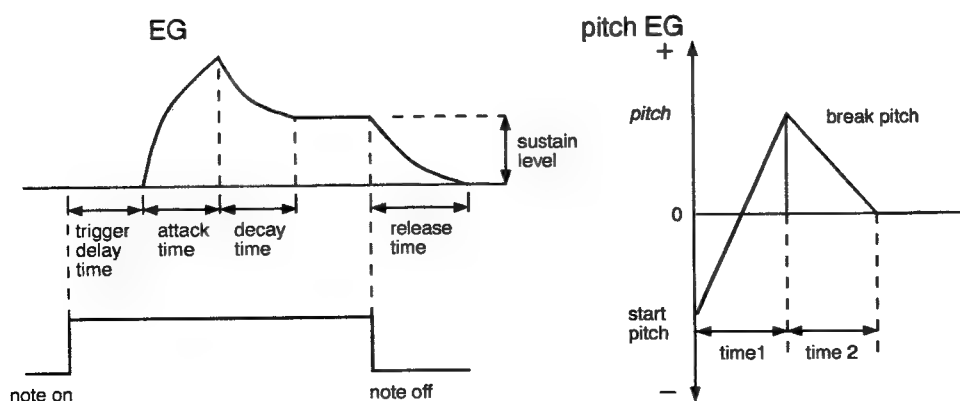


Figure 10-2 – SYN EG/pitch EG explanation

## On keyboard track

The function that changes the level and time in accordance with the note data (interval) is called *keyboard track*. The keyboard tracks for the DPS-F7 are set by the "KBD track break key" and "KBD track intensity" parameters.

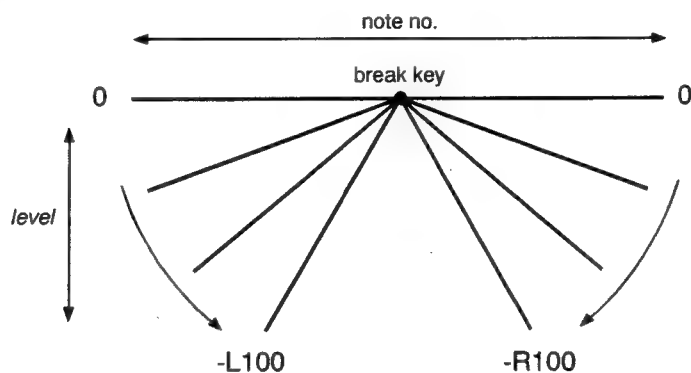


Figure 10-3 – "level KBD tracking" explanation

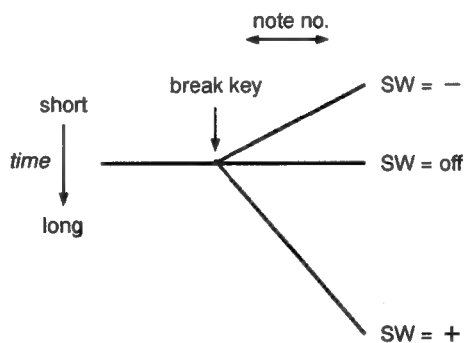


Figure 10-4 – "time KBD tracking" explanation

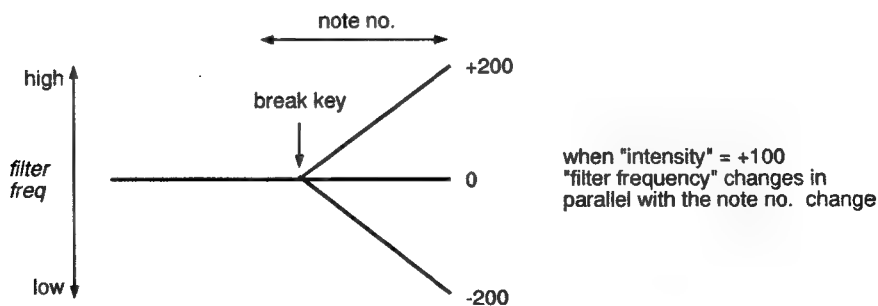


Figure 10-5 – "KBD tracking (filter)" explanation

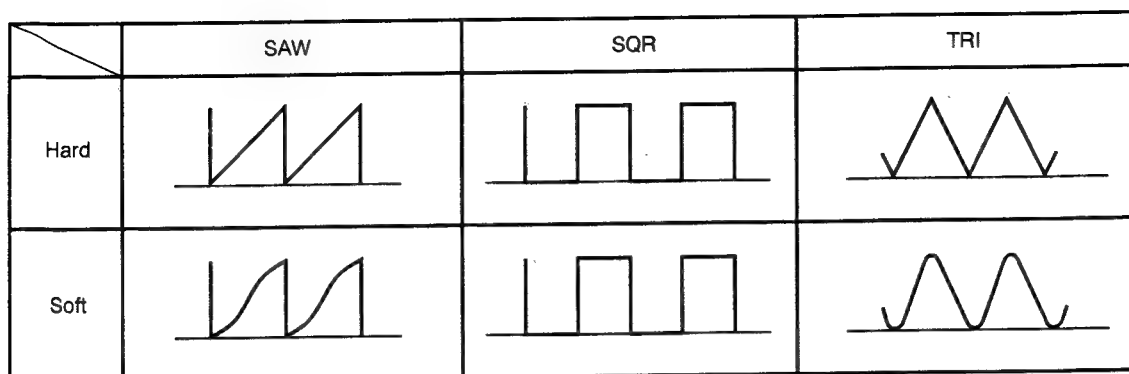
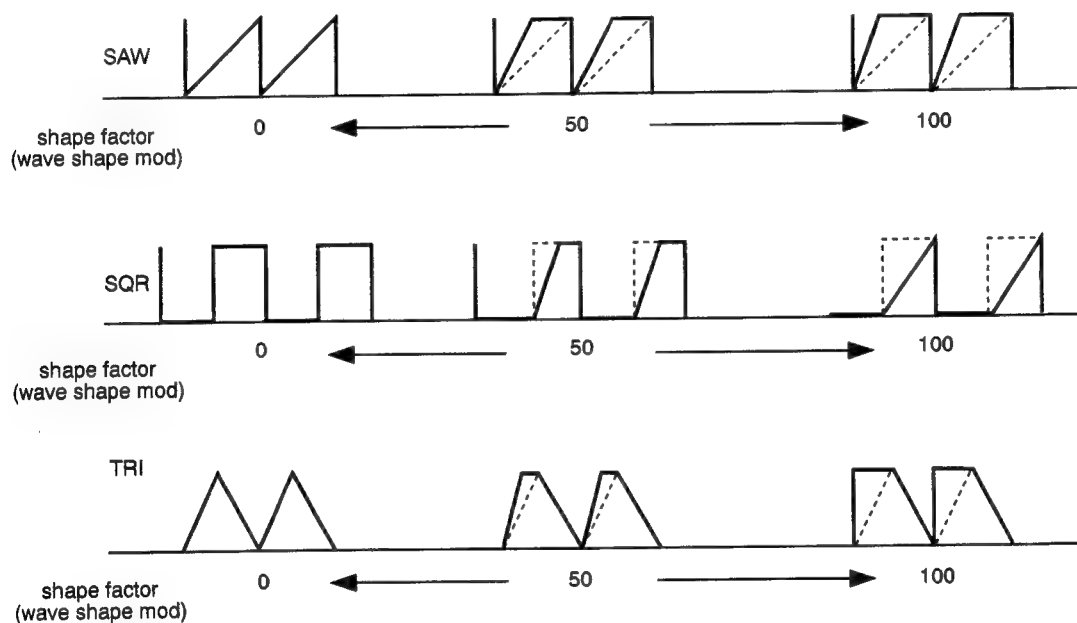


Figure 10-6 – “waveform basic/edge” explanation

**Note**

When the “shape factor (wave shape mod)” parameter has a negative value, waveform change is symmetrical at 0 as the axis of symmetry.

Figure 10-7 – “shape factor (wave shape mod)” explanation

# Other Blocks

You can use the other block than the effect block for setting the operating environments of the DPS-F7.

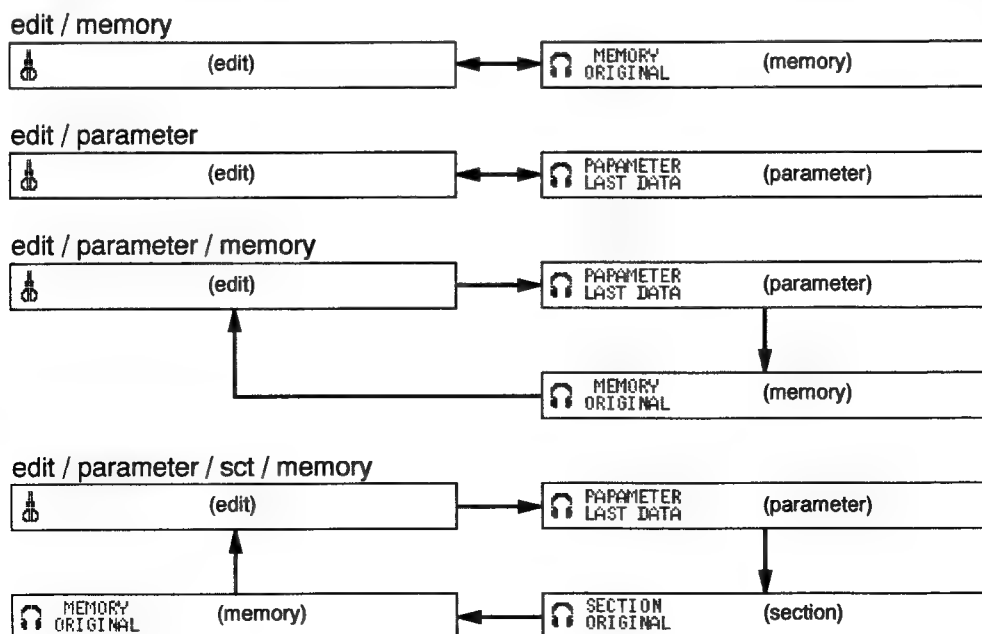
## Memory Block

This is the block for editing, protecting, and comparing the user memory.

Parameter	Meaning
memory compare	<p>For comparative listening with original memory. The following selections are available.</p> <ul style="list-style-type: none"> <li>• "edit/memory"</li> <li>• "edit/parameter"</li> <li>• "edit/parameter/memory"</li> <li>• "edit/parameter/sct/memory"</li> </ul> <p>"edit" : Normal editing mode  "parameter" : Only the currently displayed parameter has a value before changing.  "sct" : Only the currently displayed section is original  "memory" : Original data</p>
move	Moves a specified user memory to another number.
copy	Copies a specified preset memory and/or user memory to another number.
delete	Deletes a specified user memory.
exchange	Exchanges specified user memories.
remaining area	Indicates the remaining area of the user memory.
protect U1 – U256	Turns on/off the memory protection for a specified user memory.

## Memory Comparison

The "memory compare" function allows you to compare the sound you are editing with the original sound. The following diagram shows how a comparison is performed each time you press the EDIT button while editing parameters in the effect block.



## System Block

This block specifies the operating environment of the DPS-F7.

Parameter	Meaning
input mode	<p>Selects either stereo or mono input.</p> <ul style="list-style-type: none"> <li>When it is set to "stereo," you can use either a mono-in/mono-out algorithm or a stereo-in/stereo-out algorithm. For mono-in/mono-out algorithm, you can use either channel for input; however, since some mono-in/mono-out algorithms are especially designed for channel 1, usually it is the best to use channel 1 for input.</li> <li>When it is set to "mono," the input signal to channel 2 is ignored and the signal to channel 1 is sent to the effect processor where it is processed separately using the parameters set for each channel.</li> </ul>
auto help	Selects whether the HELP message is to be displayed automatically or not.
load form	<p>Selects auto load or enter load.</p> <p>auto load – a memory is automatically called up when you dial the memory number in load mode.</p> <p>enter load – a memory is not called up until you dial the memory number in load mode and then press the ENTER button.</p>
load time	Sets the memory access time after the number is changed in auto load mode. Available range is from 200 msec to 1000 msec.
program change thru level	Sets the output level during the memory number change and its processing. The output will be muted during the memory number change when this is set to "0." (This setting, however, is ignored for changes between memories in which Algorithm PEQ is used.)
edit mode	<p>Sets the edit mode.</p> <p>The following are available.</p> <ul style="list-style-type: none"> <li>"all/section"</li> <li>"all/block"</li> <li>"sync/section"</li> <li>"sync/block"</li> </ul> <p>"all" : Selects all parameters.</p> <p>"sync" : Displays and selects parameters which allow independent settings for both channels only as "sync" parameters.</p> <p>"section" : Selects parameters of the section selected on the section selecting screen.</p> <p>"block" : Selects parameters of the block selected on the block selecting screen.</p>
dial sensitivity	Adjusts the sensitivity of the dial within the range of 1 to 12. The dial sensitivity increases as the numbers become smaller.
unit (time)	<p>Specifies the unit for time information such as delay time.</p> <p>The available units are word/msec.</p> <ul style="list-style-type: none"> <li>word represents the number of samples.</li> </ul>
unit (level)	Sets the units for level information. Select either "%" or "dB."
unit (q)	Sets the unit for q of EQ. Select either "q" or "oct."
remote ch	Sets the remote channel. Select from 1 to 15 channels.
remote baud rate	Selects the remote baud rate. Select from 9600 to 31250 bps.
clock set	Sets the calendar and time. Cursor moves at a press of the EDIT button and clock setting can be confirmed when the parameter menu is displayed.
user's name	Enter your name. Cursor moves when you press the EDIT button.
date of birth	Enter your birth date. Cursor moves when you press the EDIT button.

(to be continued on the next page)

## Other Blocks

("System Block" continued from the previous page)

Parameter	Meaning
key protect display	Select either "KEY PROTECT" or "DPS-F7," which will be displayed on the screen while the "key protect" is functioning.
key protect	When this parameter is on, the front panel buttons and controls no longer work. This function is to prevent misoperation by someone else. To release "key protect," press both the EDIT and ENTER buttons at the same time and turn the dial counterclockwise. (See page 55.)
synth output	If a MIDI connection has been made and if the DPS-F7 is used as an effector such as PEQ, loading of both algorithms "SYN" and "PRC" are prohibited to prevent any sound from being generated by accident.
panel key	In some algorithms, "panel key" can be selected as the input source of the envelope follower. When one of these algorithms is loaded and "panel key" is selected, the ENTER button can control the trigger-on and trigger-off.
battery check	Checks the battery for maintaining the user memory.
version check	Checks the software version.

### SYS. SG Block

The settings selected in this block are valid only when algorithm "SYN" or "PRC" is used.

Parameters	Meanings	Valid for:
master tune	Set the Interval A within the range from 435 to 445 Hz.	SYN, PRC
transpose	Transposition is possible in $\pm 1$ oct/100 cents.	SYN, PRC
main volume on/off	Turn on/off the volume control through MIDI control change no. 7.	SYN, PRC
damper on/off	Turn on/off controlling the damper through MIDI.	SYN
portamento on/off	Turn on/off the portamento through MIDI control change no. 65.	SYN
portamento time on/off	Turn on/off the portamento time through MIDI control change no. 2	SYN

**SYS. MIDI (System MIDI) Block**

This block specifies the MIDI operating environment of the DPS-F7.

Parameter	Meaning
MIDI function on/off	Turn on/off transferring and receiving all MIDI data. (This parameter is set to "on" when the power is switched on.)
MIDI omni	Sets MIDI omni on/off. MIDI data is received regardless of MIDI channels when "omni" is "on."
MIDI ch	Sets the MIDI channel. Select from 1 to 16 ch.
program change receive	Sets program change on/off.
control change receive	Sets on/off for control change (such as modulation wheel, damper, etc.). If this is off, the set value for each parameter in the SYS.SG block is ignored and becomes off.
channel pressure receive	Sets on/off for channel pressure (after-touch).
pitch bend receive	Sets on/off for pitch bend.
system exclusive receive	Sets on/off of exclusive messages except identity, and also turn on/off receiving of the bulk dump data. (This parameter is set to "off" when the power is switched on.)
bulk dump transfer	Transfers memory data and/or system information through MIDI. The following information can be transferred. <ul style="list-style-type: none"> <li>• "all" (all user memories, system information, SYS.MIDI information, SYS.SG information)</li> <li>• "all user's memory" (all user's memories)</li> <li>• "SYSTEM" (SYSTEM information)</li> <li>• "SYS.MIDI" (SYS.MIDI information)</li> <li>• "SYS.SG" (SYS.SG information)</li> <li>• "user's memory" (one specified user memory)</li> </ul>
program change no. 1 - 128	Sets the memory numbers corresponding to MIDI program change numbers 1 to 128. Select from P1 to P100, U1 to U256, and BYPASS.

**MIDI Bulk Dump Procedures**

The DPS-F7 allows you to transfer or receive MIDI bulk dump data (memory data or system information) to or from another DPS-F7 or MIDI equipment (MIDI data filer, MIDI sequencer or personal computer) which has the bulk dump function.

**Preparation**

1. Connect the MIDI OUT terminal of one unit to the MIDI IN terminal of the other unit and MIDI IN terminal of the former to the MIDI OUT terminal of the latter.
2. Set the same MIDI channel for both units.
3. Turn "on" the "MIDI function on/off" parameter in the SYS.MIDI block on this unit.

**Bulk dump transfer**

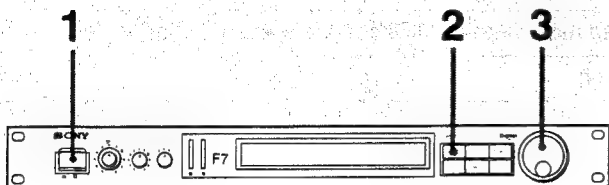
1. Have the unit of the receiving side set to receive the system exclusive message.
2. Select the data to transfer using the "bulk dump transfer" parameter in the SYS. MIDI block of this unit.
3. Press the ENTER button on this unit to start sending data.
4. After the transfer is complete, reset the receiving unit so it cannot receive the system exclusive message. It will prevent malfunctions of the other unit.

**Bulk dump receive**

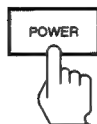
1. Turn "on" the "system exclusive receive" parameter of the SYS. MIDI block of this unit.
2. Send bulk dump data from the other unit of the transferring side.
3. After the transfer is complete, turn "off" the "system exclusive receive" parameter of the SYS. MIDI block of this unit. It will prevent malfunctions of this unit.

# Calling Up a Memory (LOAD)

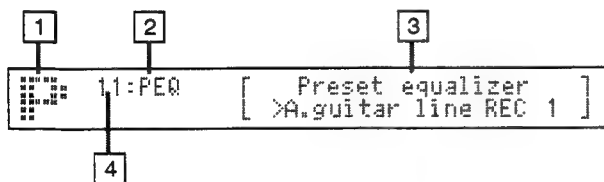
This operation calls up an effect stored in memory.



1. Turn on the power.

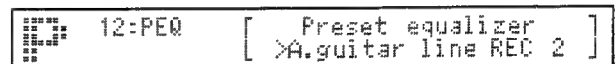


2. Press the LOAD button.



- 1 LOAD mode indication (P = Preset memory, U = User memory)
- 2 Algorithm name
- 3 Memory name
- 4 Memory number

3. Turn the dial and select a desired memory number.

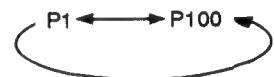


The effect of the selected memory number is automatically called up. When selecting "enter load" for "load form" in the system block, press the ENTER key after selecting the memory number. (If a number different from the effect currently called is selected, P or U will blink.)

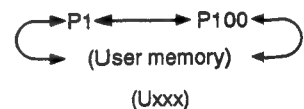
## Memory Numbers

A hundred settings are stored in the preset memory when the unit is shipped from the factory. These settings are displayed in endless order by turning the dial. If individually created settings are stored in the user memory, they will be inserted after P100 of the preset memory. After the last user defined setting stored in memory, the display sequence again starts with P1.

When shipped from factory



When stored in user memory

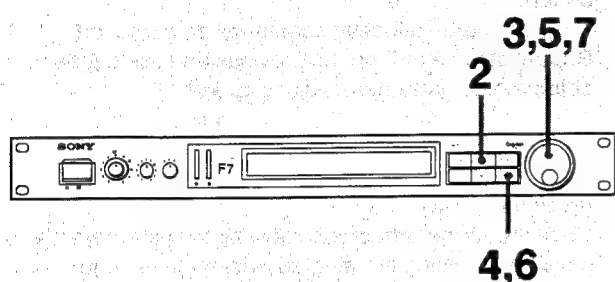


# Changing the Effect (EDIT)

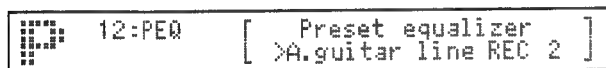
To be continued ►

This function allows you to create new effects by modifying those already stored. (See page 58 to store a changed effect, so you can use it again.) Unless a changed effect is stored, it is lost when the power is shut off.

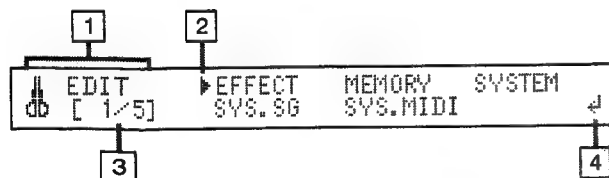
**Example: Changing the "bass frequency" parameter of "Fading Parametric Equalizer (PEQ)"**



1. Call up a memory number to be changed (see page 52).



2. Press the EDIT button so that the block selecting screen will be displayed.



- 1 EDIT mode
- 2 Displayed on left side of the currently selected item.
- 3 This shows there are 5 selections and the first of these is selected.
- 4 This means there are no more items following. If there are more, "←" appears.

3. Turn the dial and select the block to be changed.

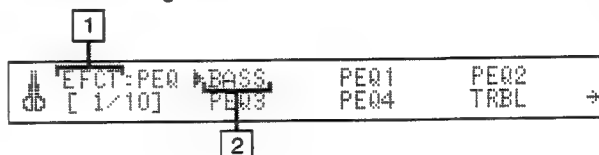


4. Press the ENTER button so that the parameter selecting screen will be displayed.

If you have selected the effect block in the above step, the section selecting screen will appear instead. So, select a section first by turning the dial and pressing the ENTER button, then go to the next step.

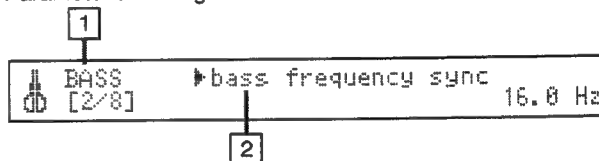


Section selecting screen



- 1 Block name
- 2 Section name

Parameter selecting screen



- 1 Section name
- 2 Parameter name

5. Turn the dial and select the parameter to be changed.



(to be continued)

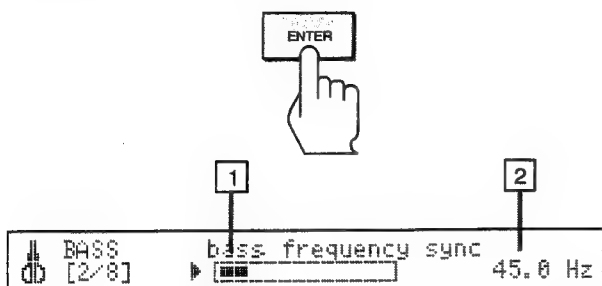
English

Calling Up a Memory (LOAD)/Changing the Effect (EDIT)

## Changing the Effect (EDIT)

(continued)

6. Press the ENTER button so that the parameter value setting screen will be displayed.



- 1 The bar graph changes according to the parameter value.
- 2 Parameter value

7. Turn the dial and change the parameter value.



### To compare the result with the former effect

Each time the EDIT button is pressed, the range of comparison is changed in the sequence set by the "memory compare" parameter of the memory block on the parameter value setting screen. By pressing the EDIT button several times, the former parameter value will resume. (See page 48.) Sounds without any effect can be produced by pressing the BYPASS button even during comparison.

### To change other parameter values in the same section

1. After changing the parameter value, press the ENTER button.  
The parameter selecting screen will be displayed.
2. Repeat steps 3 to 7 on the previous and this pages to change other parameter values as well.

### To cancel the editing and restore the former memory setting

1. Press the LOAD button.  
Once the former effect resumes, all the parameters you have been setting are deleted, with the message "Parameters have been changed. Are you sure you want to load? Yes -ENTER No -EDIT".  
If you accept deletion of the parameters being changed, press the ENTER button. Otherwise, press the EDIT button to store the effect you have created by using the SAVE function. (See page 58.)
2. Press the ENTER button.  
The former memory resumes.

### To enter the date and user information in the system block

Press the EDIT button and move the cursor.

### To change parameter values of a different section

1. Press the ENTER button after changing a parameter.  
The parameter selecting screen will be displayed.
2. Press the EDIT button (or press the ENTER button after selecting "QUIT" with the dial).  
The section selecting screen will be displayed.
3. Repeat steps 1 and 2 above to do the same for other parameters.

### What is sync?

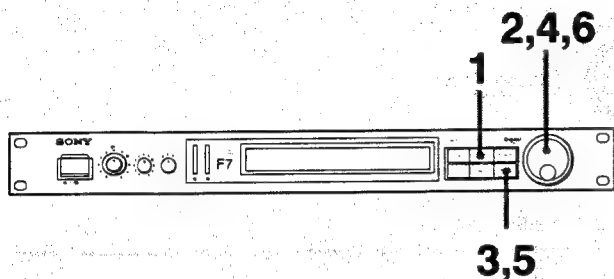
Sync is an indirect parameter which forces each channel to have the same values, even if they are set differently to start with. The values for channel 1 have priority and override the settings for channel 2, so with "sync" selected, both channels process the input signal using the settings for channel 1.

### Changing units

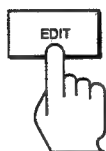
Although units are generally changed in the system block, they can also be changed in the parameter value setting screen by pressing the ENTER button while pressing the HELP button.

## Key Protection

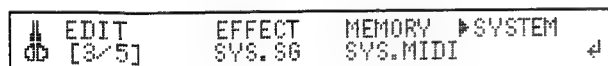
To set the "key protect" parameter



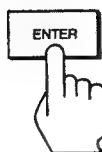
1. Press the EDIT button.



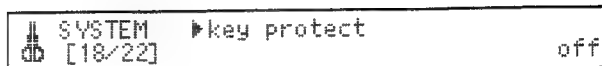
2. Turn the dial and select the system block.



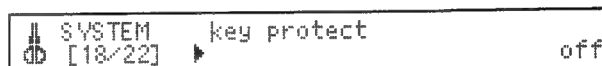
3. Press the ENTER button.



4. Turn the dial and select the "key protect" parameter.



5. Press the ENTER button again.  
The cursor moves down.



6. Turn the dial clockwise.

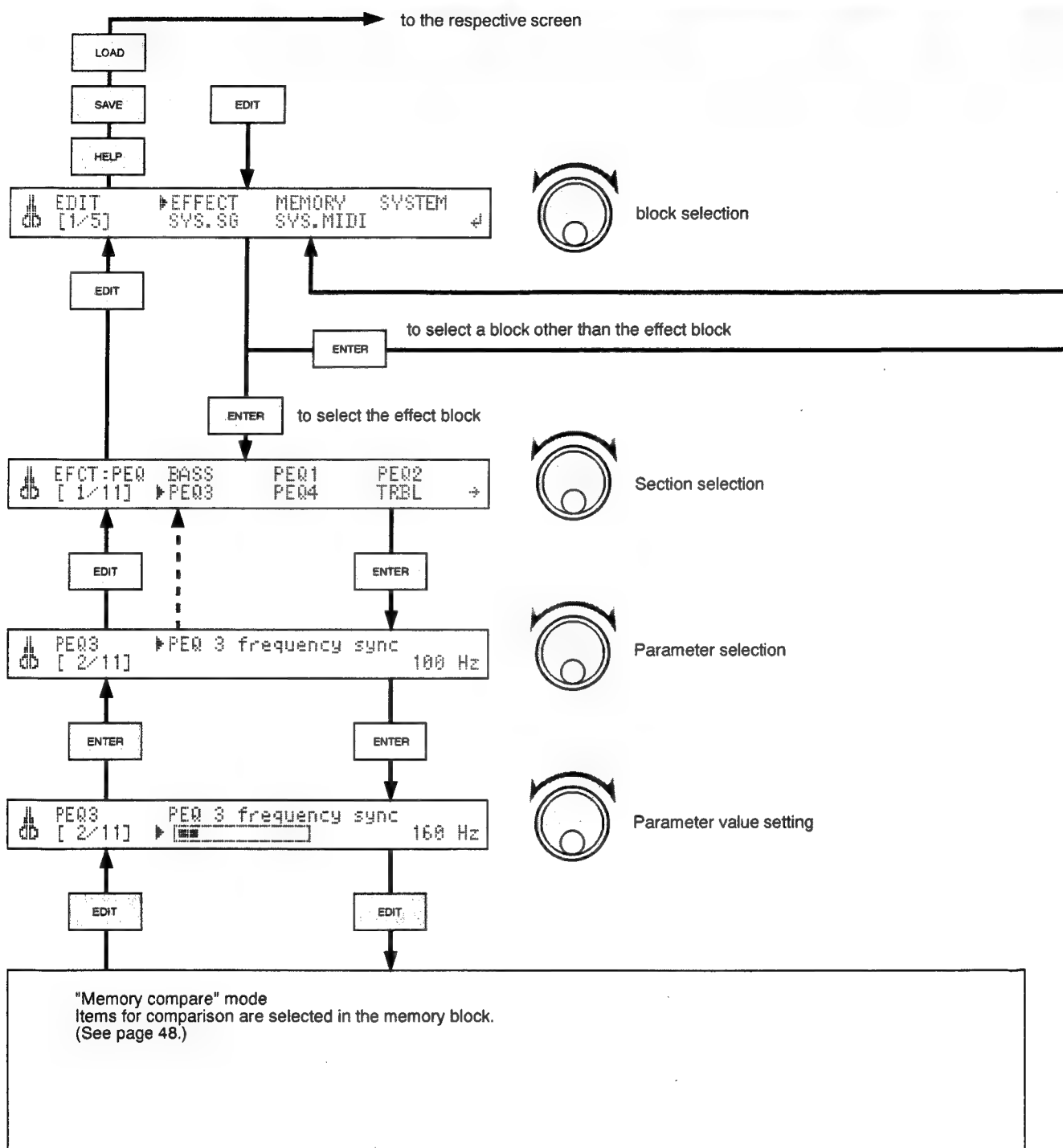


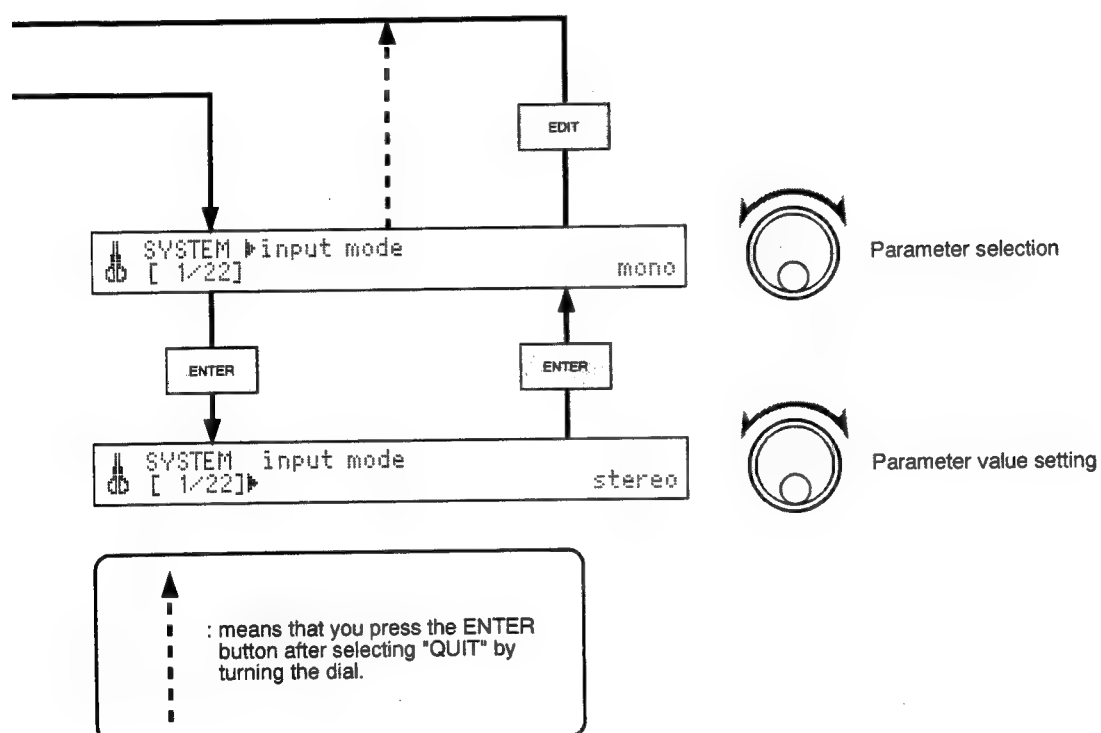
### To cancel the "key protect" parameter

Turn the dial counterclockwise while pressing both the EDIT and ENTER buttons at the same time.

## Changing the Effect (EDIT)

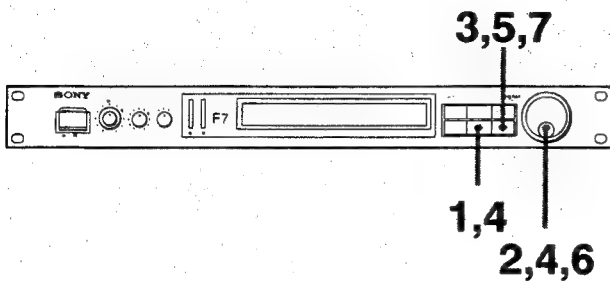
### Edit Function Flow Chart



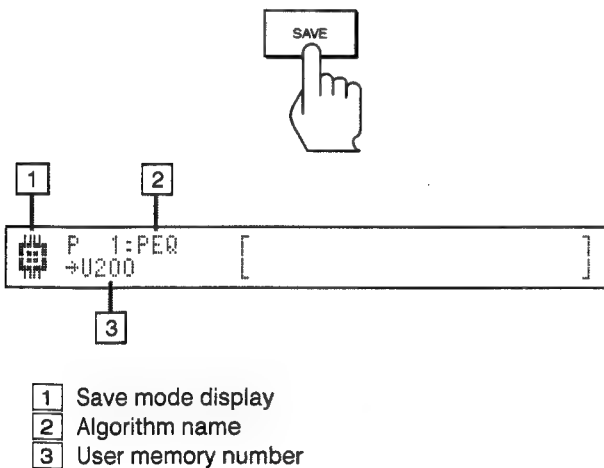


# Saving the Changed Effects (SAVE)

You can save the changed effects resulting from parameter values you have changed with the edit function.



1. Press the SAVE button.



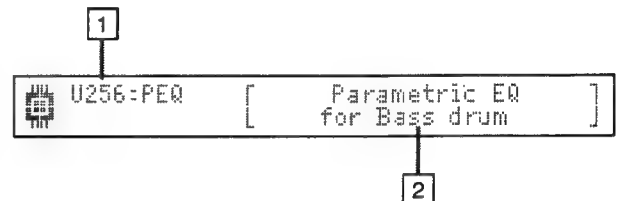
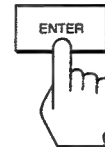
- 1 Save mode display
- 2 Algorithm name
- 3 User memory number

If you designate a memory number in which an effect is already stored, the algorithm name and memory name will be displayed after the user memory number.

2. Turn the dial and assign a number to the edited effect.



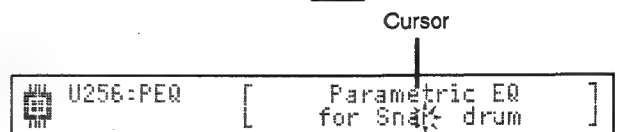
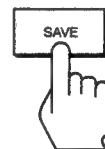
3. Press the ENTER button.



- 1 User memory number
- 2 Memory name of the original effect is displayed.

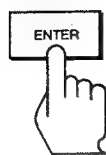
You cannot store the effect in the protected memory number indicated with "PRO" unless you release the protection. (See page 48.)

4. Denominate the memory.

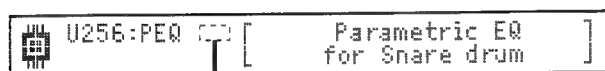


Use the dial for selecting characters and the SAVE button for cursor movement. The cursor advances each time the SAVE button is pressed. The characters are arranged in the order of 0 - 9, A - Z, symbols and dot patterns. When you want to delete the memory name of the original effect, move the cursor to the head of the memory name indication area. Select "all clear" with the dial and press the ENTER button. ("all clear" lies before "A.")

5. Press the ENTER button after denominating.



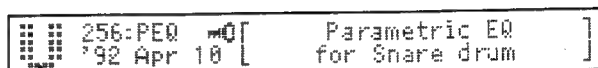
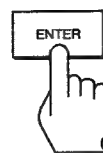
6. Turn the dial and set the memory protection if necessary.



If the memory protection is required, display the " " symbol.

Turning the dial changes on/off for the " " symbol.

7. Press the ENTER button.



When the effect is saved, " " changes to " ".

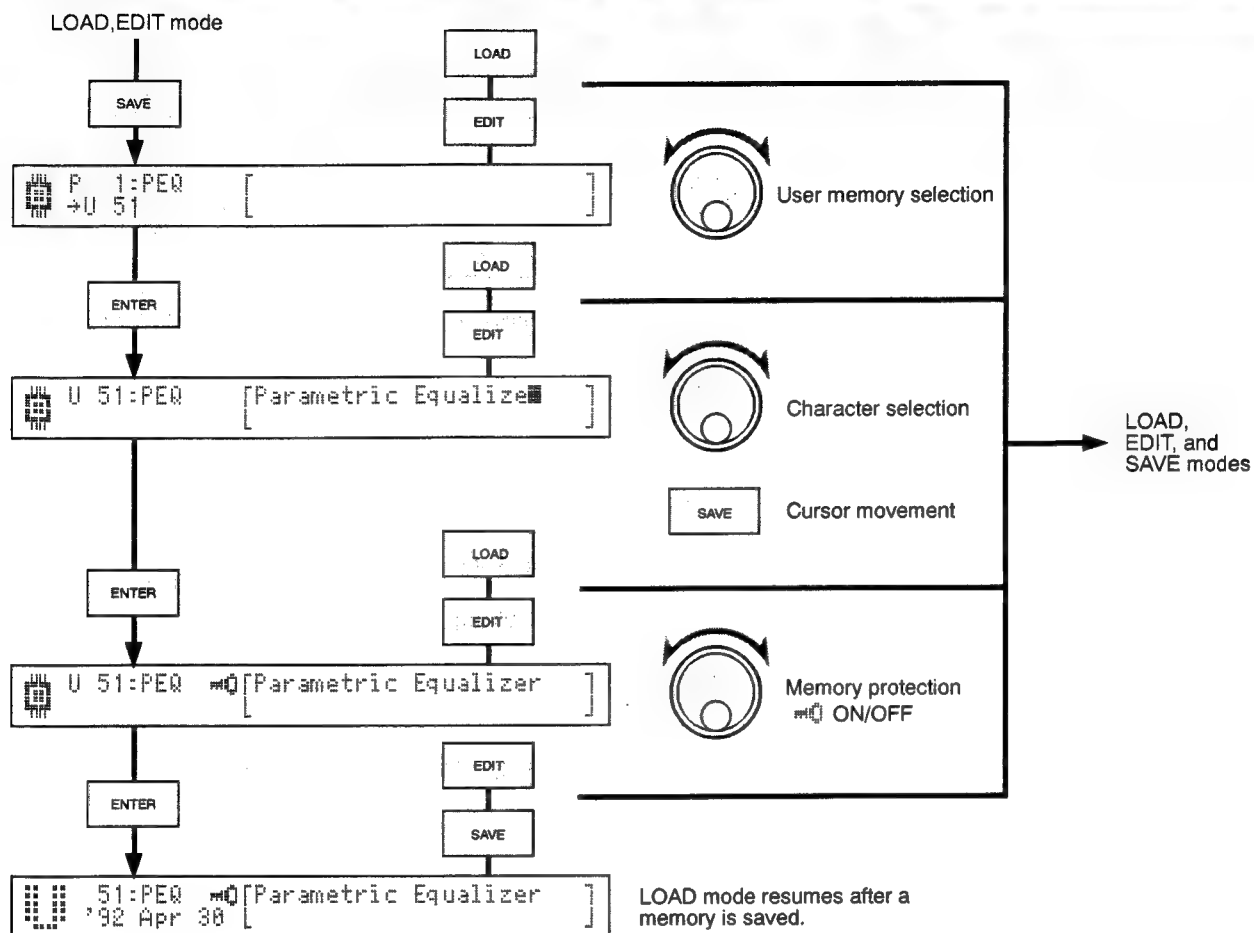
If there is not enough user memory remaining, the effect you have created may not be saved. You will see the message, "Memory full! need xxx byte more." In this case, you have to delete unnecessary memory (see the memory block on page 48) so that the memory area will have enough space to save the effect. If the effect is saved, "completed" is indicated and the SAVE mode is disengaged automatically to the LOAD mode.

#### What is memory protection?

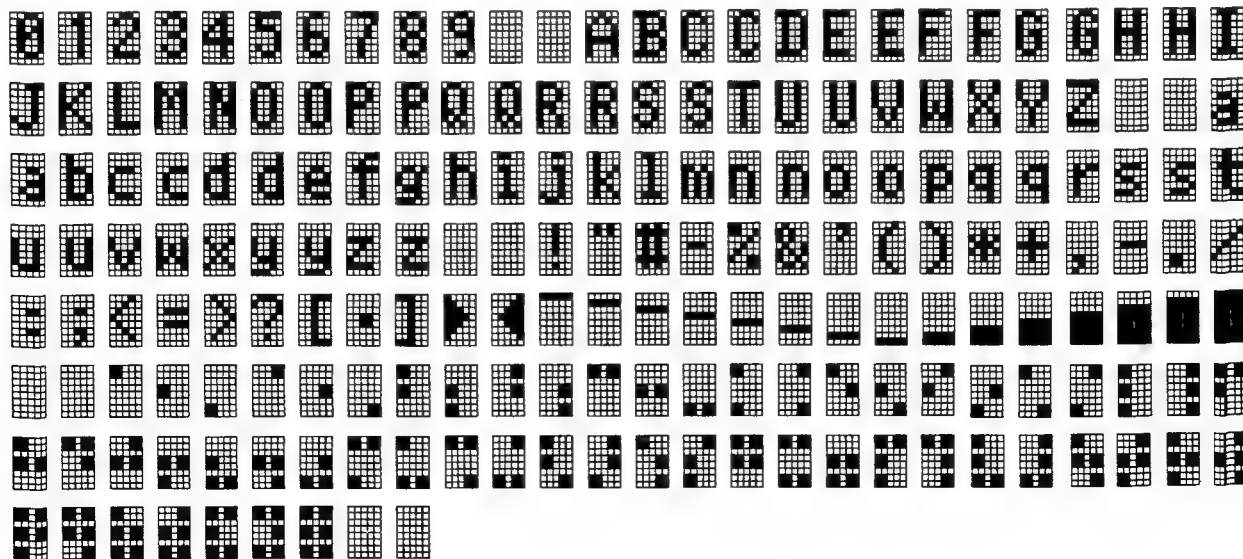
Memory protection prevents the effect you have saved from being deleted by overwriting. The protected memory number cannot be used for overwriting unless you release the protection in the memory block.

## Saving the changed effects (SAVE)

### Saving Function Flow Chart



### Available characters



When transmitting data between DPS-F7 and other devices by using MIDI, sending and receiving are possible with the data format shown below and under the conditions of the implementation chart on page 64.

**Channel voice message**

* 1 0 0 0	n n n n	Note off & channel number(n n n n = 0 - 15)
0 k k k	k k k k	Note number(k k k k k k k = 0 - 127)
* 1 1 0 0	n n n n	Program change & channel number(n n n n = 0 - 15)
0 p p p	p p p p	Program number(p p p p p p = 0 - 127)
* 1 0 1 1	n n n n	Control change & channel number(n n n n = 0 - 15)
0 c c c	c c c c	Control number(c c c c c c c = 0 - 127)
0 v v v	v v v v	Control value(v v v v v v v = 0 - 127)
* 1 0 0 1	n n n n	Note on & channel number(n n n n = 0 - 15)
0 k k k	k k k k	Note number(k k k k k k k = 0 - 127)
0 v v v	v v v v	Note on velocity(v v v v v v v = 0 - 127)
* 1 1 0 1	n n n n	Channel pressure(after-touch) & channel number(n n n n = 0 - 15)
0 v v v	v v v v	Pressure value(v v v v v v v = 0 - 127)
* 1 1 1 0	n n n n	Pitch bend & channel number(n n n n = 0 - 15)
0 v v v	v v v v	Pitch bend change LSB
0 v v v	v v v v	Pitch bend change MSB

**Channel mode message**

* 1 0 1 1	n n n n	Mode message & channel number(nnnn=0 - 15)
0 c c c	c c c c	Omni mode off      Omni mode on      All note off
		ccccccc=124      cccccccc=125      cccccccc=123
0 v v v	v v v v	vvvvvvv=0      vvvvvvv=0      vvvvvvv=0

**System exclusive message**

* 1 1 1 1	0 0 0 0 (F0)	Exclusive status	
0 1 0 0	1 1 0 0 (4C)	SONY ID	
0 0 0 0	n n n n (0n)	Global channel(nnnn=0 - 15)	} Exclusive Header
0 0 0 1	0 0 1 0 (14)	DPS-F7 ID	
0 c c c	c c c c	Command	
0 d d d	d d d d		
:			
0 d d d	d d d d	Data	
1 1 1 1	0 1 1 1 (F7)	End of Exclusive-EOX	

**ALL DATA DUMP REQUEST (Receive)**

Command : 0001 0000(10)

Data : None

**ALL USER MEMORY DUMP REQUEST (Receive)**

Command : 0001 0001(11)

Data : None

**SYSTEM DUMP REQUEST (Receive)**

Command : 0001 0010(12)

Data : None

**SYS.MIDI DUMP REQUEST (Receive)**

Command : 0001 0011(13)

Data : None

**USER MEMORY DUMP REQUEST (Receive)**

Command : 0001 010n(14or15)

bit 7

Data : 0 nnn nnnn

nnnnnnnn : User memory number-1(0 - 255)

bit 654 3210

bit 76543210

**SYS.SG DUMP REQUEST (Receive)**

Command : 0001 0111 (17)

Data : None

**ALL DATA DUMP (Send/Receive) (ALL USER MEMORY + SYSTEM + SYS.MIDI + SYS.SG)**

Command : 0001 1000 (18)

Data : 0ddd dddd .... dddddd : Data (see note 1,6)

**ALL USER MEMORY DUMP (Send/Receive)**

Command : 0001 1001 (19)

Data : 0ddd dddd .... dddddd : Data (see note 1,2)

## MIDI Implementation Chart

### SYSTEM DUMP (Send/Receive)

Command : 0001 1010(1A)

Data : 0ddd dddd....

ddddddd : Data (see note 1,2)

### SYS.MIDI DUMP (Send/Receive)

Command : 0001 1011(1B)

Data : 0ddd dddd....

ddddddd : Data (see note 1,4)

### USER MEMORY DUMP (Send/Receive)

Command : 0001 110n(1C or 1D)

bit 7

Data : 0 n n n n n n n

bit 6 5 4 3 2 1 0

: 0ddd dddd...

n n n n n n n n : User memory number-1 (0 – 255)

bit 7 6 5 4 3 2 1 0

ddddddd : Data (see note 1,5)

### SYS.SG DUMP (Send/Receive)

Command : 0001 1111(1F)

Data : 0ddd dddd...

ddddddd : Data (see note 1,7)

### START ADDRESS TRANSFER (Receive)

Command : 0010 0000(20)

Data : 0 a a a a a a a

bit 6 5 4 3 2 1 0

: 0 a a a a a a a

bit DCB A 9 8 7

: 0 0 0 a a a a a

bit 12 11 10 F E

a a a a a a a a a a a a a a a a a a : Start address (0h – 7FFFFh)

bit 12 11 10 F E D C B A 9 8 7 6 5 4 3 2 1 0

### DATA TRANSFER (Receive)

Command : 0100 0000(40)

Data : 0 a a a a a a a

bit 6 5 4 3 2 1 0

: 0 a a a a a a a

bit DCB A 9 8 7

: 0 0 0 a a a a a

bit 12 11 10 F E

: 0ddd dddd.....

a a a a a a a a a a a a a a a a a a : Start address (0h – 7FFFFh)

bit 12 11 10 F E D C B A 9 8 7 6 5 4 3 2 1 0

ddddddd : (see note 1)

### note 1- dd: Data format

0ddd dddd	0ddd dddd	0ddd dddd	0ddd dddd-
bit 765 4321	076 5432	107 6543	210 7654
← dd0	→← dd1	→← dd2	→← dd3
0ddd dddd	0ddd dddd	0ddd dddd	0ddd dddd....
bit 321 0765	432 1076	543 2107	654 3210
dd3 →←	dd4 →←	dd5 →←	dd6 →←

### note 2-ALL USER MEMORY DUMP FORMAT

dd0 ~ dd513 : USER MEMORY FAT  
dd514 ~ dd28113 : USER MEMORY DATA

### note 3-SYSTEM DUMP FORMAT

dd0 ~ dd47 : SYSTEM DATA

### note 4-SYS.MIDI DUMP FORMAT

dd0 ~ dd259 : SYS.MIDI DATA

### note 5-USER MEMORY DUMP FORMAT

dd0 ~ dd329(max) : USER MEMORY DATA

### note 6-ALL DATA DUMP FORMAT

dd0 ~ dd47 : SYSTEM DATA  
dd48 ~ dd307 : SYS.MIDI DATA  
dd308 ~ dd323 : SYS.SG DATA  
dd324 ~ dd837 : USER MEMORY FAT  
dd838 ~ dd28437 : USER MEMORY DATA

### note 7-SYS.SG DUMP FORMAT

dd0 ~ dd15 : SYS.SG DATA

## Universal system exclusive message

## INQUIRY MESSAGE

## IDENTITY REQUEST (Receive)

* 1 1 1 1	0 0 0 0	(F 0)	Exclusive status	} Universal System Exclusive Non-Real Time Header
0 1 1 1	1 1 1 0	(7 E)	Non realtime message	
0 0 0 0	n n n n	(0 n)	Global channel(nnnn=0 - 15)	
0 0 0 0	0 1 1 0	(0 6)	Inquiry message	
0 0 0 0	0 0 0 1	(0 1)	Identity request	
1 1 1 1	0 1 1 1	(F 7)	End of Exclusive - EOX	

## IDENTITY REPLY (Send)

* 1 1 1 1	0 0 0 0	(F 0)	Exclusive status	} Universal System Exclusive Non-Real Time Header
0 1 1 1	1 1 1 0	(7 E)	Non realtime message	
0 0 0 0	n n n n	(0 n)	Global channel(nnnn=0 - 15)	
0 0 0 0	0 1 1 0	(0 6)	Inquiry message	
0 0 0 0	0 0 1 0	(0 2)	Identity reply	
0 1 0 0	1 1 0 0	(4 C)	SONY ID	
0 0 0 0	0 0 0 1	(0 1)		
0 0 0 0	0 0 0 0	(0 0)		
0 0 0 0	0 1 0 0	(0 4)	DPS-F7 ID	
0 0 0 0	0 0 0 0	(0 0)		
0 s s s	s s s s	(s s)	} Software version	
0 s s s	s s s s	(s s)		
0 s s s	s s s s	(s s)		
0 s s s	s s s s	(s s)		
1 1 1 1	0 1 1 1	(F 7)	End of Exclusive - EOX	

## On RPN (Registered parameter number)

Control change no.100 and 101 have been defined as RPNs. The DPS-F7 can change parameters listed below by using these RPNs.

Designate a parameter to control with the MSB (Most significant bit) and the LSB (Least significant bit) of the RPNs. Then, set the value with data entry (control no.6 and 38)

pitch bend sensitivity	RPN	MSB = 00H, LSB = 00H
	Data entry	MSB = 00H - 0CH (0 - 12 semitones) LSB = ignored
master fine tuning	RPN	MSB = 00H, LSB = 01H
	Data entry	MSB, LSB = 33H, 2BH - 4CH, 43H (440Hz = 40H, 00H)
master course tuning	RPN	MSB = 00H, LSB = 02H
	Data entry	MSB = 34H - 4CH (-12 - +12 semitones) LSB = ignored
RPN reset	RPN	MSB = 7FH, LSB = 7FH
	Data entry	MSB, LSB = ignored

\* The "pitch bend sensitivity" parameter is valid for the "SYN" algorithm and the other parameters are valid for the "PRC" and "SYN" algorithms

**DIGITAL DYNAMIC FILTER PLUS DPS-F7**  
**MIDI Implementation Chart**

Date: Aug. 1, '92  
Version :1.0

Function ...		Transmitted	Recognized	Remarks
Basic Channel	Default	×	1 - 16	Memorized
	Changed	×	1 - 16	
Mode	Default	×	OMNI ON/OFF	Memorized
	Messages	×	×	
	Altered	*****		
Note Number :	True voice	*****	○ 0 - 127	
Velocity	Note ON	×	○9n, V=0 - 127	
	Note OFF	×	×	
After Touch	Key's	×	×	
	Ch's	×	○	
Pitch Bend		×	×	*1
Control Change	0 - 31, 64 - 120	×	○	*5 Pitch modulation *1 Portamento time *1 Data entry of LSB and MSB *4 Main volume *1, *2 Damper switch *1 Portamento switch *1 LSB and MSB of RPN *3
	1	×	○	
	5	×	○	
	6, 38	×	○	
	7	×	○	
	64	×	○	
	65	×	○	
	100, 101	×	○	
Prog Change :	True #	×	○ 0 - 127	
		*****		
System Exclusive		○	○	
Common	: Song Pos	×	×	
	: Song Sel	×	×	
	: Tune	×	×	
System Real Time	: Clock	×	×	
	: Commands	×	×	
Aux Messages	: Local ON/OFF	×	×	*1 *1
	: All Notes OFF	×	○	
	: Active Sense	×	○	
	: Reset	×	×	
Notes		*1 Reception is possible when "SYN" algorithm is used. *2 Reception is possible when "PRC" algorithm is used. *3 Reception is possible for all algorithms, but "pitch bend sensitivity" reception is possible only for the "SYN" algorithms. *4 Valid for RPN. *5 Assignment is possible in the program.		

# Specifications

A/D converter 18 bit oversampling stereo A/D converter  
 D/A converter Advanced pulse D/A converter  
 Sampling frequency 48 kHz

## Input

Connector type	Reference input level	Max. input level	Input impedance	Circuitry type
XLR-3-31 equivalent	+4 dBs	+24 dBs	10 kilohms	Balanced
Phone jack	-10 dBs	+10 dBs	50 kilohms	Unbalanced

XLR-3-31 equivalent connector (1: GND 2: HOT 3: COLD)

## Output

Connector type	Reference output level	Max. output level	Suitable load impedance	Circuitry type
XLR-3-32 equivalent	+4 dBs	+24 dBs	Over 600 ohms	Balanced
Phone jack	-10 dBs	+10 dBs	Over 10 kilohms	Unbalanced

XLR-3-32 equivalent connector (1: GND 2: HOT 3: COLD)  
 0 dB = 0.775 Vrms

## Frequency characteristics

10 Hz - 22 kHz  $\pm 0_{-1.0}$  dB  
 S/N Over 97 dB\*  
 Dynamic range Over 97 dB  
 Distortion rate Under 0.0035% (at 1 kHz)

## Memory

Preset memory 100 types  
 User memory Max. 256 types  
 Power requirement USA and Canadian model  
 120 V AC, 60 Hz  
 UK model  
 240 V AC, 50/60 Hz  
 (adjustable with a voltage selector)  
 Continental European model  
 230 V AC, 50/60 Hz  
 (adjustable with a voltage selector)

Power consumption 27 W

Dimensions Approx. 482 × 44 × 320 mm (w/h/d)  
 (19 × 1 3/4 × 12 5/8 inches)  
 (excluding projections)

Weight 5.0 kg (11 lb 1 oz)

## Accessories supplied

Power cord (1)  
 Preset memory list (1)

\* Measured at the digital full scale level

Design and specifications are subject to change without notice.

## Note:

This appliance conforms with EEC Directive 87/308/EEC regarding interference suppression.

# Troubleshooting

Symptom	Check if:
Power does not go on	• The power cord is plugged into the outlet.
No sound	• The INPUT control is set to "0." • The OUTPUT control is set to "0."
No sound effect	• The bypass circuitry is functioning.
Sound is distorted	• Input level is too high. → Lower the input level by turning the INPUT control counterclockwise.
No stereo effect	• "input mode" of the system block is set to "mono."
Uncontrollable with MIDI	• MIDI receive channel is suitable for sending channel of the MIDI device. • The control number assigned to this unit is used. • "MIDI function on/off" of the SYS. MIDI block is set to "on."

**Parameter**

A number of elements are involved in creating each effect. One effect is obtained only after determining the values of the elements required. Each of these elements is called a parameter.

**Indirect parameter**

This is a parameter that can be changed according to preset rules while editing. "sync" is a typical example.

This is not an actual parameter (parameter that can be saved) but is a convenient parameter that can change the multiple parameters.

**Memory**

This is an internal memory circuit. The DPS-F7 has a built-in microcomputer that sends the set value of each parameter to the signal processing LSI (DSP) to create the various effects. If the data of this parameter is stored in the memory, it can be retrieved and used when needed.

The DPS-F7 has 100 preset memories (memory initially set at time of shipment) and a maximum of 256 user memory (memories that are available to the user).

**Temporary buffer**

This is a place where the parameter of each effect is loaded and edited. Each effect is reproduced by the parameters called into this temporary buffer.

**Load**

Calling the effects stored in the memory is called "to load." The parameters stored in the preset memory and user memory are copied in the temporary buffer and then new parameters are reflected in the DSP.

**Edit**

Changing the value of a parameter is called "to edit," and individual effects can be created by changing values of parameters in the temporary buffer. This function is to make the effects in the preset memory more effective by conforming with usage conditions and the user's own tastes.

**Save**

Storing parameters in the temporary buffer as user memory is called "to save" and is an important function to store individual effects. Individual effects once saved can be freely accessed for editing and saving again.

**MIDI**

This is the abbreviation for Musical Instrument Digital Interface and is an international standard for data communication between electronic musical instruments. This enables automatic performance by controlling other musical instruments from one keyboard or by using a sequencer and computer. The MIDI function of the DPS-F7 enables selection of memory numbers with MIDI program change numbers (tone quality change signal from the keyboard) and control of parameters by means of the MIDI control change signal (amount of change of the modulation wheel and so on).

**Algorithm**

A fundamental arithmetic method is required in the internal circuit of the digital dynamic filter plus to obtain an effect and different arithmetic methods are used such as for dynamic filter effect, parametric equalizer effect and synthesizer effect. Any one of these arithmetic methods is called algorithm. Great many numbers of newly developed algorithm are incorporated in the DPS-F7 for variegated effects far exceeding those available with conventional effectors.

# Block Diagram

